EXECUTIVE SUMMARY: Voice over IP is one of the fastest growing technologies and in just a few years will be in 80% of all companies. As companies look to replace their existing analog phone systems with VoIP systems, the unique requirements of conference rooms need to be considered. If they are not specifically addressed, it will result in poor audio quality leading to frustrated users, missed conversations, and potentially lost business.

This white paper will provide IT managers with insights into the impact that room acoustics and the variety of AV and user devices have on the quality of the user experience.
It’s Wednesday morning and time for the weekly status update with your company’s largest customer. After 30 minutes of crowding around the conference phone struggling to hear what the customer is saying—and worrying you aren’t being heard—the stress levels start to rise. At the one-hour mark, it’s surprising how tired everyone is. Trying to decipher clipped speech and ignore that constant echo is exhausting.

This scenario plays out on a daily basis in the corporate workplace. Traditionally, it would fall to an audio visual (AV) expert to resolve these types of quality issues. However, as telephone systems move from analog phone lines to the IP network, these issues become the responsibility of corporate IT groups. IT managers are being tasked with purchasing and supporting Voice over IP (VoIP) communication systems for the company, including the conference room.

This responsibility comes with three important goals:

• Create a high quality audio experience for end users that minimizes distractions like echoes and ambient noise and that is easy to use.
• Ensure the highest voice quality of all remote participants, regardless whether they are on a cell phone, landline, or VoIP line.
• Make certain that all of the AV components in the room can be used with the new VoIP system.

To meet these new challenges, you must develop a solid understanding of audio needs and room design. You should also plan for and invest in the right equipment. This white paper provides a framework for evaluating VoIP conferencing solutions with an introduction to audio and room design principles. It also offers recommendations for audio conferencing solutions that integrate the IP network with various user devices to produce a high quality user experience.

Advancements in VoIP networks, endpoint devices, and protocols—as well as the drop in price for high-speed access—have resulted in a mass migration to VoIP systems. According to recent research by In-Stat, VoIP penetration is expected to reach 79% by 2013. That means almost 8 out of every 10 companies will rely on VoIP for their communications. VoIP adoption is growing at a rapid pace—if your company hasn’t begun implementing VoIP, you can expect it will soon.
VoIP has the potential to take voice communication to the next level. While there are significant cost savings and plenty of features that come with VoIP systems, the real advantage is the voice quality. Not only has VoIP call quality matched Public-Switched Telephone Network (PSTN) quality, it gives companies the opportunity to far surpass PSTN with wideband audio.²

Voice is the most critical component of any communication system because it creates a personal connection between people. The ability to enhance voice clarity and ensure every nuance is heard and understood can make the difference between a successful sales pitch to an international customer or a frustrating exchange that ends in misunderstanding. If you are tasked with providing a quality user experience, this level of clarity—where every word is heard and conversations flow naturally—is the desired outcome that VoIP can help make a reality.

The emphasis on voice clarity does not minimize other benefits you can create for your company by switching to a VoIP system. VoIP helps IT achieve business objectives such as agility, cost control, and scalability. You can easily add users, change extensions, move phones, and save on long-distance charges and phone lines.

For these reasons, analysts agree that VoIP will continue to see tremendous growth. This growth places more expectation and responsibility on the IT manager. And, in order to be successful, you need to have a complete understanding of the options available for conference rooms.

A 2008 study by Minacom showed that enterprise and residential VoIP quality surpassed PSTN voice quality, with an average Mean Opinion Score (MOS) of 4.2, compared to 3.9 for the PSTN.²

A LOOK AT AUDIO BASICS

As an IT leader, the network is your domain. Now that audio resides on the network, you are responsible for creating productive user experiences in conference rooms and ensuring that all audio signals, regardless of source, are seamlessly and clearly integrated into audio and video conferences.

Audio visual integrators spend decades learning the complexities of audio processing and room acoustics. While it is not required that you develop that same level of expertise, you’ll benefit from a basic understanding of audio concepts. This introduction to room considerations and audio terminology will help you plan and invest in a conferencing solution that will create successful, productive meetings with any remote participant.
**ROOM ACOUSTICS AND USE**

Room acoustics refer to how sound behaves based on certain characteristics of the room such as surfaces and size. A room with a glass wall or with a number of hard, reflective surfaces will change how the audio sounds. It may reverberate or sound hollow, reducing intelligibility. Wall treatments and audio processing equipment can combat quality issues created by hard surfaces and glass walls.

Larger rooms require more loudspeakers to fill the room with sound so all participants can hear. They also require more microphones to pick up all participants’ voices so they can be heard by remote participants. In some very large rooms, it may be necessary to also reinforce local audio through loudspeakers so that people in the back of the room can hear. Generally speaking, the larger the meeting room, the more requirements there are that must be met to deliver a natural experience.

Another important consideration is the purpose of the room. Will it be used strictly for teleconferencing or will there be additional audio sources from DVD players, video conferencing units, and even PCs? If your conference room needs to accommodate multiple remote parties at the same time, it is important to ensure all audio is fully integrated so all participants can hear each other clearly regardless of whether they are connecting by cell phone, video conference, landline, or PC.

**Figure 1.** Conference systems need to integrate all types of endpoints and allow all participants to communicate clearly with each other.
**Audio Concepts**

| **Acoustic Echo Cancellation (AEC).** | When audio from the remote caller is picked up by the microphones in your conference room and sent back to the caller, it creates echo. Echo is extremely distracting and creates listener fatigue—if the participants are even willing to hang in that long. Echo is eliminated using a technique called acoustic echo cancellation. Basically, a conferencing system with acoustic echo cancellation has the ability to compare audio signals being sent and received and eliminate duplicate signals or echo. This is a critical feature for any conferencing system. |
| **Ambient Noise.** | Ambient noise is the background or room noise that occurs in every space. These noises compete with voice audio and can make it difficult to understand what is being said. Noise cancellation technology reduces ambient noise so that only voice audio is transmitted to remote meeting participants instead of the hum of air conditioning, the sound of pages being turned, coughing, or other such background noises. |
| **Audio Bandwidth.** | Bandwidth in the audio world refers to the frequency range of the audio signal rather than speed or size. Wideband audio is 7kHz, which provides fuller, more natural sound. By contrast, narrowband audio, or that of a standard telephone, is 3.3kHz. |
| **Digital Signal Processing (DSP).** | DSP refers to various techniques used to enhance the clarity and intelligibility of audio before it is transmitted to the remote party. This is particularly important when you have a wide mix of audio sources. |

**Extending VoIP to the Conference Room**

“Participants shouldn’t be aware of the technology that enables such an immersive experience... The overarching goal is to support continuous interaction—even over several hours—without inducing participation fatigue.”

—Frost & Sullivan

With a basic understanding of audio concepts and room consideration, you can more effectively evaluate your options. Extending VoIP to the conference room can be achieved in a few different ways. However, these options do not provide equal results and not all will meet the demands for an enhanced user experience, a consideration that should be paramount. As younger people enter the workforce, their expectations for ease of use and quality are naturally high. They are, after all, “native users” of technology and want their devices at work to at least measure up to their devices at home. Executives also demand high quality audio that will make them sound good to customers and other high profile contacts.
VoIP gives you the opportunity to meet these expectations by bringing higher quality, more natural conversations to the conference room along with wideband audio capability. And that’s where the conferencing system matters. No matter how great VoIP could make someone sound, it can only be as good as that user’s equipment. If the equipment only supports narrowband audio or can’t adjust for room acoustics and ambient noise, the call quality is impacted. Keep in mind that to achieve wideband quality, a wideband CODEC must be used on the other end of the call.

The right audio conferencing system delivers on the promises of VoIP and ultimately results in more productive meetings, accommodates more participants, and enhances the company’s image. Simply put, you’ll sound better to the people you call.

There are several options for bringing VoIP to the conference room. While there are valid reasons for selecting each, their limitations must still be considered.

**IP CONFERENCE PHONES**
Plug-and-play devices are always appealing because they are so easy to set up. IP conference phones are typically provided by the IP experts who consulted on the VoIP desktop phones. But here is where the gap between IT and AV comes into play. While these phones meet IT requirements by being an easy addition to the network, they cannot provide the highest quality audio and meetings are limited to very few participants. However, when you consider room acoustics, room size, other multimedia components in the room, and management and control, IP conference phones don’t measure up. That’s not to say they don’t have their place; they can be suitable for small rooms that are used solely for conference calls.

**ANALOG TELEPHONE ADAPTERS**
Another option, particularly for companies that have existing audio conferencing equipment, is an analog telephone adapter (ATA). An adapter does meet the basic requirement of allowing analog devices to connect to a VoIP network, but it is not an optimal solution. In addition to losing VoIP call quality, it creates added cost—both in hardware and programming.

“Telepresence can deliver a remarkable communications and collaboration experience, but to do so, companies must plan for and invest in certain key components for success. Those include video and audio technology.”

—Frost & Sullivan
PROFESSIONAL VOIP-ENABLED AUDIO CONFERENCING SYSTEMS

Just as installed audio systems are superior to tabletop units when using analog lines, the same is true in a VoIP system—and for many of the same reasons. Professional audio conferencing systems help companies achieve superior voice clarity by compensating for challenging room acoustics and applying digital signal processing to all audio signals. Installed systems tie together all room components that handle audio, such as DVRs, video conferencing units and computers. It can also be linked to a room controller for ease of use.

Quality is just one of the many advantages of an installed audio conferencing solution versus using an IP conferencing phone. Additional benefits include:

- Full-duplex speech transmission
- Improved sound in large and acoustically challenging rooms
- Elimination of echoes and ambient noise
- Ability to accommodate large numbers of meeting participants
- Integration of VoIP conference call with other AV devices in the room and video codecs

In addition to room requirements and audio capabilities, it is important to find out how the VoIP conferencing system interacts with your network. AV integrators can help you determine the right VoIP solution by bringing years of experience in audio system design as well as an understanding of the top network questions you need answered. Together, you can create a professional audio conferencing solution that delivers the VoIP voice clarity and ease of use that your users expect. To make sure a VoIP conferencing solution will meet your network and room requirements, ask the following questions:

**NETWORK REQUIREMENTS**

- Is it standards-compliant? For most VoIP conferencing systems, your IP PBX needs to be SIP compliant or have a third-party SIP converter.
- Does it require a Proxy Server? It is likely you will need a proxy server in the VoIP system, such as the Cisco® Unified Communications Manager (CallManager).
• Do you need additional licenses? An endpoint user license may be required for the audio conferencing system, just as for any third-party endpoint or telephone.

• Can you configure Quality of Service (QoS) settings? Find out what the configuration options are and where that information is entered.

ROOM REQUIREMENTS

• How does the system accommodate challenging room acoustics? Make sure the system can provide acoustic echo cancellation, noise cancellation, and digital signal processing.

• How does it work with other audio sources? Can it integrate all room audio sources such as DVR players, computers and video codecs?

• Can it bridge the audio from different endpoints (e.g. video conferencing, cell phones, landline, and desktop conferencing) and enable all participants to hear each other?

• Is it scalable? Will it meet your current and future needs?

MAXIMIZING VOIP WITH BIAMP® SYSTEMS

Through the Audia® series of professional audio conferencing products, Biamp Systems takes audio quality far beyond what’s common for IP conference phones or analog telephone adapters. AudiaFLEX enables you to deliver high quality audio conferencing experiences to your users. When regional sales teams call into their weekly sales meeting, AudiaFLEX ensures every sales person will be able to hear and be heard in natural, full sound. Virtual sales meetings—or any type of distance conference—will become more productive and interactive.

AudiaFLEX also gives you ultimate flexibility in design. You can add features and functionality to match your specific room requirements. Choose any configuration of input and output pairs to accommodate all your audio signals. Inputs for a typical conference room could include audio from local room microphones, remote participants, DVR players, computers, and video codecs. Outputs could be used to send audio to loudspeakers in the room as well as to conferencing participants.
The AudiaFLEX AEC-2HD card adds acoustic echo cancellation to significantly improve call clarity, delivering more natural sound during distance conferences.

You can conference over VoIP directly from AudiaFLEX by adding the VoIP-2 card, which uses the SIP message protocol. Up to six VoIP-2 cards can be installed into a single AudiaFLEX unit for a total of 12 VoIP lines. Additional AudiaFLEX units can be added to a system to increase the total number of VoIP lines.

The AudiaFLEX conferencing system with VoIP-2 also has the unique ability to bridge the audio between video codecs and teleconferences. This capability enables remote participants to clearly hear and be heard by other remote participants as well as by the participants in the local conference room. For example, participants connecting to the meeting via video conference will be able to interact with the person calling in on a cell phone and the person calling in on a landline for more natural conversations—as though they were all in the same room.
THE BIAMP ADVANTAGE FOR IT

The Audia system makes it easy to take on the VoIP challenge for conference rooms and create an enhanced user experience. It provides:

• Digital signal processing for all audio sources—Ensures audio is clean and clear before transmission. AudiaFLEX allows all AV components to interface with the VoIP conference.

• SONA™ AEC—This acoustic echo cancellation algorithm improves voice quality in any conference room for real, full sound.

• Networked control—Facilitates deployment, maintenance and management, and can be controlled remotely. IT administrators can quickly change settings or simply unmute a line from any location.

• Flexible Meeting Space—AudiaFLEX with VoIP-2 cards is flexible enough to support multiple room configurations and room combining. For example, systems can be configured to easily go from four conference calls in four rooms to one conference call in a large, combined room.

• Bridging—Each VoIP-2 card supports two lines for a total of 12 lines for each AudiaFLEX. Simply add more AudiaFLEX units for more VoIP lines. This capability enables enterprises to easily create their own bridges rather than paying for a conferencing service. It also allows you to tie a VoIP teleconference to a video conference so all participants can hear each other.

• Standards compliance—The VoIP-2 card uses the SIP message protocol.

From boardrooms with distance conferencing to large group meeting rooms, Biamp products offer you the freedom and flexibility to design feature-rich audio systems that evolve to meet changing business requirements.
VoIP will soon be the standard in corporate communication, with nearly an 80% adoption rate expected in just the next couple of years. Because VoIP can support wideband audio that is more than double that of PSTN, users have the potential to experience a much more natural sound and increased intelligibility.

The benefits of VoIP will naturally extend to the conference room. Users have come to expect high-level sound quality, without clipping or frustrating ambient noise—something IP conferencing phones can’t provide. As an IT leader, it’s important to reach out to your team and educate them on the benefits of a professional audio conferencing solution.

With the AudiaFLEX solution from Biamp Systems, you can eliminate listener fatigue caused by muffled audio on conference calls. Your teams will sound great in their high-profile customer meetings and weekly status updates. And that dreaded Wednesday call? Well, it’s likely to be a lot shorter without the constant interruption to ask “What did you say?”

**CONCLUSION**


Maximizing the Value of Telepresence Interoperability is Key to Success, Frost & Sullivan

Ibid
With years of VoIP experience, Biamp Systems continues to deliver innovative, quality conferencing solutions for the enterprise. To learn more about professional VoIP conferencing systems, room considerations and network requirements, visit the Biamp Systems web site at www.biamp.com. You’ll find a number of VoIP resources including learning modules, configuration questionnaires, and tech notes.