Hear your customers loud and clear with wideband (HD Voice) technology

White paper

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Introduction

Wideband or HD Voice is an important area of IP communications currently receiving a lot of attention and from where there have been key rollout announcements such as recently, from France Telecom/Orange[1].

Although many of the announcements have been in the mobile communications space, HD Voice has the potential to bring huge improvements to enterprise VoIP platforms such as conference servers, call centre servers, and unified communications systems. HD Voice promises to bring improved call clarity and a better user experience to voice communications. It is easy to see why the extra clarity of communication would be important to scenarios such as air traffic control or emergency services operator/dispatcher calls. However, it is equally important for any business operating in today's competitive environment to give the best possible customer service, and with HD Voice capability built into your voice platforms your customers will appreciate your investment.

The technology behind HD Voice has been around for quite some time, however, it is only in the last few years that it has gained the momentum needed to succeed and become a ubiquitous technology. Early rollouts created wideband islands; however we are finally seeing some movements within the industry that will alleviate this problem, thus giving the promise of high quality calls to everyone.

This whitepaper discusses the technology behind HD Voice and sets out why it is an important feature to consider when choosing your next communications platform.

Factors impacting the voice communications platform industry today and challenges to overcome

Increasing globalisation

In recent years, many service organisations such as mobile phone companies or banking institutions that utilise call centres have chosen, due to economic considerations, to use call centre operations based in a different country to their own. Whilst this was an economic success, it also led to increased customer dis-satisfaction[2] caused by poor communications technology.

The voice communications in use today are based on traditional telephony standards that haven't changed much since the 1950s. These standards were set at that time and limit the information bandwidth for voice communications to 300-3400Hz (200-3200Hz in the US and Japan). However, if one analyses normal conversational speech, it typically covers the frequency range 0-8000Hz and the human ear is capable of hearing frequencies up to 18 or 20kHz. Back when these standards were set, it was felt that a voice channel limited to 3.4kHz would be good enough and ever since that time we have all accepted that telephone conversations would have a slightly muffled tone.

The public switched telephony network (PSTN) in widespread use today still utilises these mature standards, but IP-based voice communications (VoIP) have now evolved to a state where there is technically no reason, on a VoIP call, to limit the clarity of the call by restricting its audio bandwidth.

Environmental concerns

A further global trend where HD Voice technology can offer benefits comes from the current recession and global warming concerns, both of which are contributing to an increased requirement and usage of audio and video conferencing facilities.

Videoconferencing systems have been evolving steadily since their introduction in the late 1960s. Digital telephony standardisation and the use of ISDN in the 1980s brought their adoption to a wider audience than before with the possibility of a 128kbit/s link. However, the evolution from TDM to IP-based networks has meant that much larger bit-pipes became available and economically feasible, thus leading to today's high-definition video conferencing systems. These high-end systems, from companies such as Polycom, Tandberg and Cisco, have high quality video and audio to give the best possible experience. Their disadvantage is that to give the best quality video and audio they require expensive, specially equipped rooms.

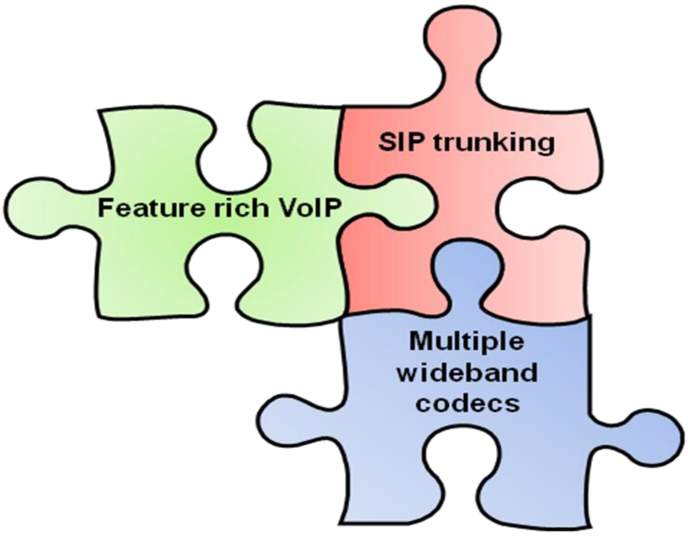
With the adoption of IP telephony within enterprises, desktop-based audio conferencing is gaining momentum, either to collaborate with colleagues around the globe, or to conference with customers. However, audio conferencing is predominantly narrowband, and is therefore not always seen as an ideal solution for important customer meetings. This has meant that it has not achieved its full potential to save money for businesses.

Cost-efficient platforms

There is an over-riding move towards VoIP-based systems in both enterprise and carrier communications, with one reason being the promise of lower OPEX. However, the migration has been slower than many predicted, and the current economic climate is forcing many enterprises to evaluate any purchase decisions much more closely. Any new IP-based system therefore needs to prove its worth and have significant benefits over the TDM counterpart being replaced.

Wideband (HD) voice technology is the solution

The pieces of the puzzle are now coming together to enable wideband IP-based (VoIP) communications to become ubiquitous:



VoIP with true feature benefits over PSTN, not just cheaper

SIP trunking technology and providers to enable interconnection of VoIP 'islands'

Multiple choices available for the voice codec to meet the needs of all user types (mobile, fixed, consumer, enterprise)

Wideband voice requires that communications are fully IP-based; however, we are still currently in the transition phase from a TDM to an IP-based voice world. For anyone specifying or designing a new voice platform today, the platform needs to have support for a myriad of features only possible or economically feasible on IP-based systems (such as IM integration, video support, virtual hosting, easy scalability and network connectivity).

The ideal solution for a new voice system would be a flexible platform with support for a wide array of both narrowband and wideband voice codecs, and support for both TDM and

IP, such that migration from TDM to IP could be achieved over time without drastic network changes being required.

A wideband voice platform offers all the benefits of VoIP platforms that have been available for some time, with the key advantage that it gives much improved voice quality, and this should be yet another tipping point in favour of VoIP.

Perhaps one of the main reasons why wideband voice has taken so long to proliferate comes from the truism that for any communications service to be successful there must be others with similar devices to call. It is fairly straightforward to create, for example, within your own enterprise, a wideband 'island' where internal calls can be made using full wideband quality, but the problem to date has been how to connect up these wideband islands. SIP trunking with wideband support between islands is the solution, along with interconnect agreements between SIP trunking providers. This means you can have some confidence that when you try to place your wideband call, a wideband voice capable IP link to the destination will be available.

There is currently a large selection of wideband codecs available from which to choose. Some, such as G.722, are more prevalent than others[3], but there does not at this stage appear to be any one codec that will become the de-facto standard. Additionally, both devices on a call need to support the same wideband codec, otherwise the call cannot be set up in wideband mode. It therefore follows that technology used at the core of a wideband voice platform should support as many wideband codecs as possible, otherwise the scope of supported wideband end point devices is unnecessarily reduced.

With multiple wideband codecs available on the core platform, transcoding is required to connect together endpoints using dissimilar wideband codecs. Transcoding requirements will impact on the overall channel throughput of any system, as it uses CPU resources to adjust from one standard to another. It also introduces additional latency into the call, which has to be managed closely to avoid impacting quality. However, in the present circumstance, where it is the case that there is no single wideband codec suited to all requirements, transcoding is necessary, and the communications platform should be able to support it whilst providing a large number of calls per server.

A further trait of any platform being considered for deployment should be the ease with which it can accept upgrades, for example, to add further wideband codec support. It should be straightforward to perform a software upgrade on the platform to add further wideband codecs as the market develops.

The clarity improvements made possible by HD Voice technology would give huge improvements across all voice communications platforms. For companies using overseas call centres it would result in improved customer satisfaction scores. However, audio conferencing is perhaps the area where the most can be gained from implementing HD Voice support, and there are certainly signs recently that this is the case – Wydevoice, TurboBridge and ZipDX are just some examples from an ever growing list of vendors with wideband conferencing products or services.

What to look for in an HD Voice platform

The key considerations when looking for an HD Voice capable platform are:

- Which codec to choose
- Flexibility
- How to integrate the new VoIP platform with the legacy TDM platform

Requirement 1 – codec choices

G.722.2 (AMR-WB) G.722 G.729.1

> "The nice thing about standards is that there are so many of them to choose from."
> (Andrew S Tanenbaum)

G.722.1 G.711.1 SILK iSAC G.718 Siren7 BV32 Speex

There are many possible VoIP codecs to choose from, each with specific attributes to make them suited to particular applications. A useful comparison can be found in Polycom's whitepaper[4].

It has been found that for voice applications, most of the benefit comes from extending the bandwidth out to 7kHz. Standards bodies such as the ITU-T[5] and developers of voice applications have chosen, on the whole, to develop applications supporting 'wideband' as opposed to 'super-wideband' or 'fullband' codecs. The latter have applications in high-end conferencing systems to convey 'presence' information and for music but would be overkill for most voice applications.

The most common wideband codecs currently deployed or supported by widely available endpoints are the original G.722 codec (standardised in the 1980s), the G.722.2 (AMR-WB codec) used in mobile applications, and codecs such as iSAC from GIPS, or Skype's SILK.

In developing your solution, it should be remembered that when a call is made from one device to another, a negotiation of the codec to use for the conversation is undertaken, with each device supporting a list of codecs, wideband and narrowband. For a wideband call to be made from device A to B, the **same** wideband codec must be supported in both devices. If the two devices do not have any wideband codec in common, then the conversation will be set up using a narrowband codec, likely G.711, that most if not all devices will support. Care must, therefore, be exercised in choosing the wideband codec to deploy. A wise choice would be to make sure the voice platform chosen can support a number of the most commonly used codecs, not just one. In that way, there is more chance that the call can be set up in wideband mode, thus allowing all the benefits of wideband voice to be maintained.

Requirement 2 – platform flexibility

IP-based voice communications, and wideband voice technology are still evolving and codec choice is constantly expanding. Consideration should therefore be made as to how easy a platform is to upgrade; ideally if a new codec or feature becomes popular then it should be a matter of a simple software upgrade to get the new functionality at minimal cost.

Additionally, any choice of voice platform should also be able to adapt to support other technologies that may be required in the future. This could be the flexibility to use different signalling mechanisms such as H.323 or SIP, or it could be the flexibility to allow expansion to higher call volumes than originally intended.

Requirement 3 – integrating the VoIP and TDM platforms

Whilst wideband voice does require fully IP-based connections, in a large majority of cases the new VoIP platform will be deployed alongside an existing legacy TDM platform, and there will be a handover period where users are ported across. It is therefore useful to have a VoIP platform that provides simple integration with TDM. A platform that can combine TDM and IP communications on the same hardware and software is ideal, as explained in the example below.

In the case of an audio conferencing platform, the configuration, if it was deployed today, it might have support for 90% narrowband lines and 10% wideband, as that ratio is what can be expected for users connecting to the platform. Over time, the ratio will change to support a higher percentage of wideband connected users. Having separate platforms for TDM and IP-based conferencing means that initially the TDM platform would be highly utilised and the wideband IP platform under utilised. Over time, the TDM platform will become more redundant and the IP platform more utilised. If a common platform were used, with a common core for the mixing of the audio streams in a conference session, then it is relatively easy to re-allocate resources from TDM to IP and narrowband to wideband as needed.

Summary

The time is right, and the technology is ready for HD Voice to be successfully deployed in enterprise and service provider voice platforms. By choosing a platform with multiple codec support the benefits of wideband calls can be extended to the widest possible audience. A platform that can support integration with legacy TDM infrastructure from a vendor willing and able to add further key functionality as the market evolves will allow a successful migration to wideband VoIP for any business.

References

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About Aculab

Aculab is an innovative, market leading company that provides world class IP and media processing boards and software to the global communications market. With many years of experience in helping to drive our customers' success, our enabling technology provides the essential components required to deliver multimodal voice, data, fax and video solutions for use within IP, PSTN and mobile networks – with performance levels that are second to none.

Aculab serves the evolving needs of developers, integrators, service providers and equipment manufacturers with cost-effective, deployment proven, high performance products. Companies worldwide have adopted our technology for a wide variety of business critical services and solutions.

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