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Application Note

Enabling HD Voice Support in Dialogic® Host Media Processing (HMP) Software Release 4.1LIN

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Application Note

Executive Summary

The next-generation of voice quality for telephony audio, commonly known as HD Voice, has arrived with the advent of wideband audio codecs. By using wideband audio connections, HD Voice is able to more accurately reproduce the human voice, enabling users to experience more natural sounding speech over a phone line. On IP packet networks, the benefit of HD Voice is that its bandwidth requirement, due to voice data compression using wideband audio codecs, can be similar to the bandwidth requirements of traditional telephony networks whose narrowband audio codecs they are replacing.

Dialogic® Host Media Processing (HMP) Software is well positioned to scale with the current and next-generation multi-core processors, providing a cost-effective platform to enable HD Voice. The use of Dialogic® HMP Software devices and their corresponding Dialogic® HMP Software libraries supporting HD Voice allow developers to enable the next-generation of voice services in their applications.

This application note focuses on the emergence of HD Voice, the use of wideband codecs, and how application developers can enable HD Voice in their Dialogic HMP Software applications.

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Application Note

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Introduction

This application note focuses on the emergence of HD Voice, a term used to describe the next-generation voice quality for telephony audio. HD Voice enables a wider range of frequencies for human voice characteristics compared to standard “toll quality” audio. On IP packet networks, an advantage HD Voice has – due to its voice data compression using wideband audio codecs – is that its bandwidth requirement can be specified to be similar to the bandwidth requirements of traditional “narrowband” telephony networks.

This application note also discusses HD Voice enabled services and how application developers can enable HD Voice in their Dialogic® HMP Software applications, specifically using the Dialogic® Host Media Processing Software Release 4.1LIN, which is well positioned to scale with current and next-generation multi-core processors, providing a cost-effective platform to enable HD Voice.

HD Voice in Dialogic® HMP Software

What is HD Voice?

HD Voice refers to the next-generation of voice quality for telephony audio resulting in high definition voice quality compared to standard digital telephony “toll quality”. HD Voice uses wideband codecs (such as G.722, G.722.2) audio connections to more accurately reproduce the human voice with a wider range of frequency coverage. The result is significantly more natural sounding speech and a wider range of sounds promoting audio clarity and clear conversation.

HD Voice is a significant step in the evolution of audio clarity and quality for telephony systems, one which can lead to greater customer satisfaction. In comparison between HD Voice and traditional telephony audio, many people can distinctly hear a difference and the general sentiment is that HD Voice provides more of a feeling of “being in the same room” with the person on the other end of the phone line.

One reason new users experience such a marked improvement in quality with HD Voice is that traditional telephony is constrained by decades old standards. Digital telephony standards (for example, ITU-T G.711) are based on 1960s digital circuit technology and 1930s microphone technology. Until the advent of HD Voice, G.711 was the standard for quality, with mobile telephony typically providing less than G.711 quality because of the bandwidth constraints within wireless networks.

Wideband Speech Codecs

In the telephony world, speech audio is sampled, digitized, and compressed. This process acts as a band pass filter to encode data within a specific audible frequency range. Traditional timeslot-based audio is sampled at 8 kHz to provide an audio frequency range of approximately 200-3400 Hz. This frequency range was considered acceptable in providing the majority of the voice energy in normal speech communication over the phone, while eliminating high frequency noise. However, the true human speech range includes sounds well above 3400 Hz and as high as 18 kHz. What is lost from traditional audio encoded data are the nuances of speech that help clarify the tones at both the very low and high side of the audio spectrum. Timeslot-based Pulse Code Modulation (PCM) encoded voice data provides this frequency range at 8 bits/sample and requires a bandwidth of 64 kbps. Other narrowband codecs, such as the AMR-NB codec used for GSM mobile networks, achieve this general frequency output at a lower bandwidth of up to 12.2 kbps, resulting in more complexity in the compression algorithm.

HD Voice is enabled through the use of wideband audio codecs, which are sampled at a higher rate of 16 kHz to provide an audio frequency range of approximately 50-7000 Hz. The higher sampling rate means that the wideband audio codecs can reach almost double the frequency range that is audible using narrowband codecs. The wider frequency range in turn enables the speech to be clearer and crisper, capturing the natural inflections in people’s voices that often peak above or below the traditional narrowband audio codec standards. While wideband codecs often use more bandwidth to represent the greater frequency range, they were also standardized to provide bandwidth rates that are comparable to narrowband rates, so that a wideband version can be used in place of its narrowband predecessor.

G.722

G.722 is an ADPCM codec that was standardized by the International Telecommunication Union (ITU) in 1988 at 48 kbps, 56 kbps, and 64 kbps rates. At 64 kbps, G.722 specifically uses 48 kbps to encode the lower <4000 Hz frequency range and 16 kbps to encode the higher 4-7000 Hz frequency range. This two sub-band sampling allows G.722 at 64 kbps to cover the 50-7000 Hz audio frequency range as a replacement for the traditional G.711 PCM encoded data.

G.722.1

G.722.1 is a second standardization for wideband speech, comparable to G.722 at the target bitrates of 24 kbps and 32 kbps. The G.722.1 codec works in low bitrate environments with reasonably low frame loss.

AMR-WB (G.722.2)

AMR-WB was standardized first for mobile GSM networks by the 3rd Generation Partnership Project (3GPP). AMR-WB was also standardized by the ITU as G.722.2. AMR-WB is targeted for wideband speech at bitrates between 12 kbps and 24 kbps. Like its narrowband predecessor, AMR-NB, the AMR-WB codec consists of several modes aimed at providing error concealment and error resiliency in error-prone mobile networks. The AMR-WB modes consist of bitrates, 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85, and 6.6 kbps. All the AMR-WB modes provide wideband audio frequencies, with rates of 12.65 or higher used under normal conditions, and lower rates used for audio stability in times of poor network conditions.

HD Voice Enabled Services with Dialogic® HMP Software

HD Voice Enabled Endpoints

HD Voice is enabled in IP and mobile networks to enhance the audio communication quality while maintaining the same general bandwidth envelope.

In fixed Voice over IP networks, G.722 is normally the codec of choice because it is an ITU standardized royalty-free codec that can be used to replace the G.711 codec without using more bandwidth.

In error-prone mobile networks, the AMR family of codecs has been adopted by the 3GPP for GSM networks because of the ability of the AMR codecs to provide excellent error concealment and error resiliency. The AMR-WB (G.722.2) codec can be used at similar 12-13 kbps bandwidth as the standard AMR-NB codec, but provide a much wider range of speech fidelity.

HD Voice Applications

The addition of HD Voice to Dialogic® Host Media Processing Software Release 4.1LIN allows developers to enable the next-generation of voice services in their applications. Dialogic HMP Software 4.1 allows developers to HD Voice enable traditional voice applications, such as Interactive Voice Response (IVR), Messaging Servers, Call Center, and Audio Conferencing applications. HD Voice is also available for many multimedia applications, such as video portals, video messaging services, video gateways, and video conferencing applications.

Dialogic® HMP Software HD Voice Wideband Audio Codecs

Dialogic HMP Software 4.1 supports G.722 and AMR-WB (G.722.2) codecs over Real-time Transport Protocol (RTP) for HD Voice, enabling IP packet network voice or multimedia applications. Dialogic's support for AMR-WB over 3G-324M networks to 3G mobile video handsets is targeted for future HMP releases.

Dialogic® HMP Software HD Voice Architecture

The Dialogic HMP Software architecture was updated to support the coexistence of both wideband and narrowband endpoints in the same applications. This was achieved by a Dialogic HMP Software architecture that has the infrastructure to support the higher sampling rate required by HD Voice wideband codecs (normally 16 kHz or higher), as well as infrastructure to support the 8 kHz sampling of traditional timeslot-based linear PCM telephony systems. This HMP architecture allows a supported HD Voice enabled endpoint to be connected to another traditional narrowband audio endpoint that is supported by HMP.

Supporting HD Voice and Narrowband Voice

To support both wideband and narrowband endpoints in the same system, the HMP architecture uses a sampling rate conversion to up-sample the voice data to 16 kHz or to down-sample it to 8 kHz.

If an HD Voice endpoint is connected to a narrowband endpoint, the voice data is down-sampled from the HD Voice 16 kHz endpoint and converted into the common 8 k linear PCM format. Hence, HD Voice data can be connected to a traditional endpoint, but the wider range of voice frequencies is lost.

In the reverse direction, narrowband voice data is up-sampled from 8 kHz to 16 kHz to reach the HD Voice endpoint. This translates to the 8 kHz voice data being sampled at twice the rate, and transmitted using 16 kHz fidelity, but it also means the missing frequency range cannot be recovered. Therefore, the voice can be encoded with high fidelity, but still only contain the same frequency range as in the original narrowband voice signal.

Within the HMP architecture, an HD Voice enabled HMP device can work with both wideband and narrowband codecs. HMP devices that are HD Voice enabled to support 16 kHz voice data can maintain the HD audio fidelity when connected to other HD Voice enabled HMP devices. Traditional voice and TDM devices can remain as narrowband 8 kHz devices, but are not precluded from connecting to HD Voice enabled devices through sampling rate conversion of the audio data.

Enabling HD Voice in Dialogic® HMP Software

Enabling HD Voice in Dialogic HMP Software involves using HD Voice enabled Dialogic HMP Software devices and their corresponding Dialogic HMP Software libraries. This section of this application note focuses on which Dialogic HMP Software libraries support HD Voice, as well as how Dialogic HMP Software devices are interconnected to achieve common usage scenarios, and lastly provides significant device details for enabling HD Voice in an HMP application. Some of the example application scenarios discussed include: HD Voice Streaming over RTP, HD Voice Play and Record, HD Voice Conferencing, HD Voice for 3G Video Calls, and DTMF generation and detection in HD Voice Calls. Scenarios also show common HD Voice use cases by connecting the appropriate HMP devices.

Dialogic® HMP Software Libraries Supporting HD Voice

HD Voice is supported by certain Dialogic HMP Software libraries and Dialogic HMP Software devices. These HD Voice enabled software devices support the internal audio port connection architecture. The internal audio port connection is a packet-based connection, similar to RTP, that can support the higher sampling rates required by HD Voice sampling. In contrast, the traditional timeslot-based devices (and connections) are not enabled for HD Voice sampling rates.

Table 1 lists the Dialogic HMP Software libraries and software devices that enable HD Voice applications.

Dialogic HMP Software Library	HMP Software Device	Functionality
Global Call IP Call Control	IPT device	Supports negotiation of HD Voice wideband codecs in 3PCC mode
IP Media Library	IPM device	Supports streaming of HD Voice wideband codecs over RTP
Multimedia API Library	MM device	Supports HD Voice play/record with HD Voice files
CNF Conferencing API Library	MCX device	Supports HD Voice Conferencing; HD Voice and/or narrowband audio mixing
3G-324M API Library	M3G device	Support for HD Voice via AMR-WB audio to/from 3G-324M endpoints is targeted for a future release. Refer to the HMP Release Updates for notification of availability of this feature.
Device Management API Library	"dev_API"	Supports HD Voice audio connections between HD Voice enabled devices
Voice API Library	DX device	Supports inband DTMF detection from and generation to HD Voice endpoints. Narrowband audio play/record (**Note:** dx_voice functions are narrowband audio functions that can be used for play/record to wideband endpoints, but do not maintain HD Voice fidelity)

Table 1. Dialogic® HMP Software HD Voice Enabled Libraries and Software Devices

HD Voice RTP Streaming

SIP Negotiation of HD Voice Codecs

SIP session negotiation for HD Voice wideband codecs is supported by the Dialogic® Global Call API in 3rd Party Call Control (3PCC) mode. In 3PCC mode, an application is responsible for:

- Parsing the Session Description Protocol (SDP) in the SIP messages to negotiate media codecs
- Independently starting the appropriate IP Media Streaming (IPM) device for RTP media streaming

Note: HD Voice wideband codecs are not supported by the Dialogic Global Call API in 1st Party Call Control (1PCC) mode.

For examples of SDP parsing, an application developer can use the SDP API convenience library and parsing code located in the ~/dialogic/demos/sdpapi/ directory. The Multimedia Demo provides an example of how to use 3PCC and the SDP API convenience library for SDP parsing of wideband codecs.

IP Media Streaming (IPM) Device

RTP Media Streaming with HD Voice wideband codecs is supported by the Dialogic® IP Media API and the IP Media Streaming (IPM) device. To start RTP streaming of an HD Voice codec, an application calls ipm_StartMedia() with the appropriate eCoderType value in the IPM_AUDIO_CODER_INFO structure.

For G.722, add the coder type values CODER_TYPE_G722_64K and CODER_TYPE_G722_64K_NATIVE.

For AMR-WB, the coder type values are added to support the various AMR-WB bitrates, so that the application can specify a specific AMR-WB rate. The values added have the form CODER_TYPE_AMRWB_xx_xxK and CODER_TYPE_AMRWB_xx_xxK_NATIVE, where xx_xx specifies one of the AMR-WB bitrates (23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85, and 6.6). A coder type set to CODER_TYPE_AMRWB_NONE allows inband negotiation of bitrate through the AMR-WB standard.

IPM HD Voice RTP Streaming Use Cases

The IPM device supports transcoding to 16 k linear PCM for connection to other HD Voice enabled endpoints and end-to-end HD Voice application scenarios. To preserve HD Voice wideband connections to other HD Voice enabled devices, connections are made with the Dialogic® Device Management APIs, dev_Connect() or dev_PortConnect() API.

When connected to narrowband endpoints, such as those using G.711, G.729, G.723, or AMR-NB codecs, the IPM device uses sampling rate conversion between devices. Figure 1 shows how transcoding is accomplished between a wideband HD Voice IP endpoint and a narrowband IP endpoint when the dev_PortConnect() API is used between IPM devices.

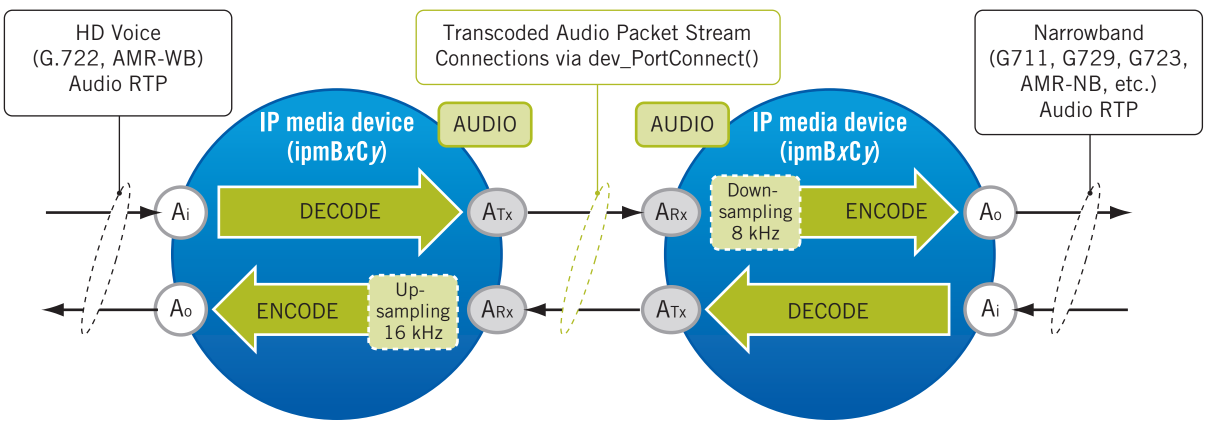

Figure 1. HD Voice Sampling Rate Conversion between IP Endpoints

The IPM device also supports native RTP streaming of HD Voice wideband audio codecs for applications that do not require audio transcoding. The coder types with _NATIVE should be used for native streaming or when no transcoding is required. A coder type without _NATIVE can be used with or without transcoding, but requires a coder license per ipm device in order to use the specified codecs. Figure 2 shows how two HD Voice IP endpoints can be connected without transcoding if both endpoints use the same wideband codec.

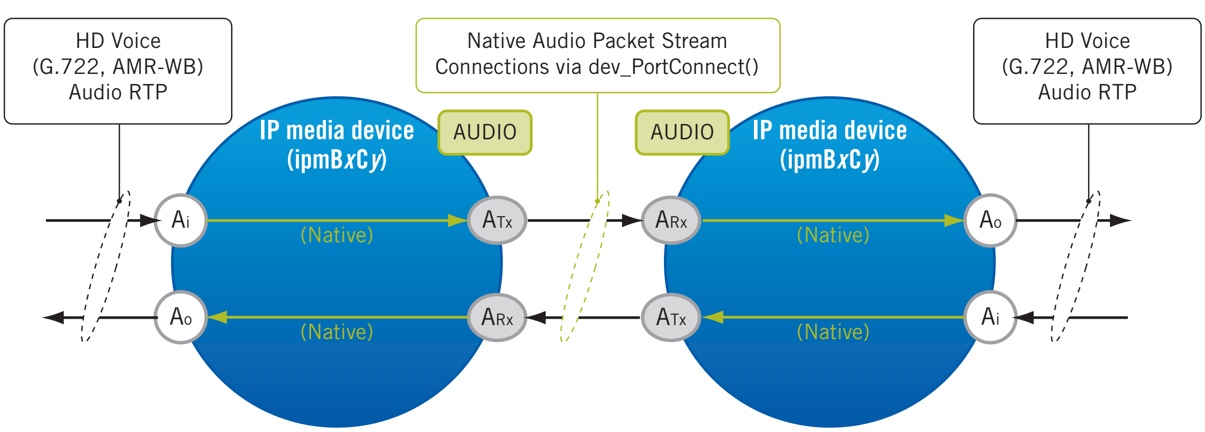

Figure 2. HD Voice Native Wideband Connection

The IPM voice data also can be routed to a 8 kHz timeslot-enabled device, such as voice(DX), narrowband audio conferencing (CNF), or E1/T1 PSTN timeslot (DTI) by using the xxx_listen() and xxx_unlisten() API. Since these timeslot-based devices only support 8 k linear PCM connections, the HD Voice wideband audio is down-sampled to 8 k linear PCM format and the internal connection is confined to the narrowband frequencies. Figure 3 shows how an HD Voice IP endpoint would be connected to an E1/T1 PSTN timeslot via a DTI device. In this scenario, the CTBus transmit timeslots are connected by using the ipm_listen() and dti_listen() APIs.

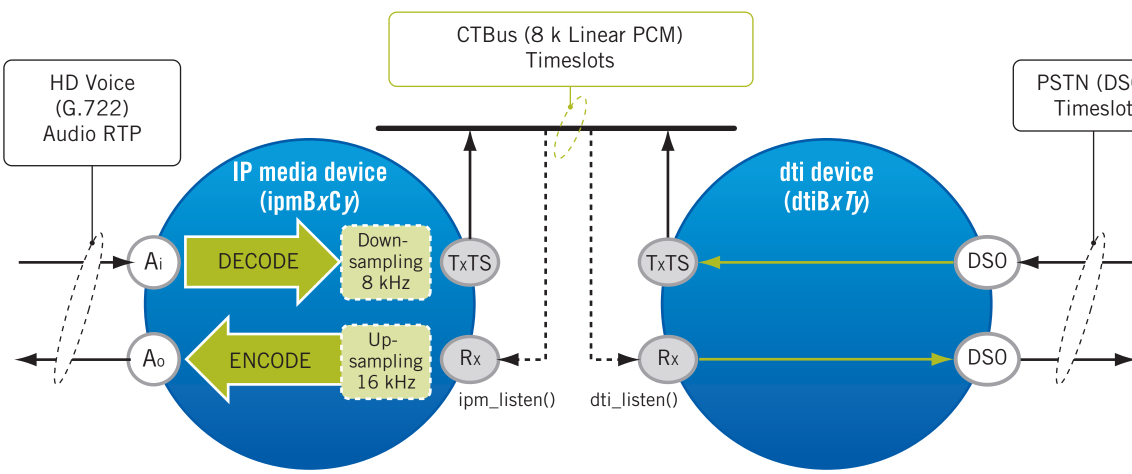

Figure 3. HD Voice to PSTN Timeslot Connection

HD Voice Play and Record

Multimedia (MM) Device

Play and Record of HD Voice wideband audio codecs is supported by the Dialogic® Multimedia API and the Multimedia (MM) device. The Multimedia (MM) device is available in two licensable options that support HD Voice and narrowband:

- A full multimedia MM device for audio+video play/record
- An audio only MM device for play/record of wideband and narrowband audio without the video functionality

To support HD Voice audio fidelity, the MM device is connected to another HMP device with the Dialogic® Device Management API, dev_PortConnect() or dev_Connect() API (for more information, see "Connecting Dialogic® HMP Software HD Voice Enabled Devices"). The dev_PortConnect() and dev_Connect() API audio connections support wideband audio fidelity through an internal packet-based connection. With transcoding enabled, these connections can provide sampling rate conversion between the network format and the mm file format selected by the application. In the case of an mm_Play, the audio format specified in the mm file can be converted to the network format and up-sampled or down-sampled, if necessary. In the mm_Record direction, the format chosen by the application for the mm file determines if the network data will be up-sampled or down-sampled as it is stored.

MM HD Voice Play/Record Storage Formats

To provide HD Voice fidelity to endpoints, the audio format of the file provides 16 kHz sampling. To accommodate this, the Dialogic® Multimedia API supports play/record of 16 k linear PCM (.pcm) and 16 k linear Wave (.wav) files for audio transcoding applications, as well as Dialogic® Proprietary file (.aud) formats for G.722 and AMR-WB for native play/record applications. To support HD Voice multimedia applications, Dialogic® Multimedia API support for 3GP (.3gp) files with an AMR-WB audio track is planned.

MM HD Voice Play/Record Use Cases

Figure 4 shows basic HD Voice media flow for Play/Record from an MM device to an HD Voice wideband IP endpoint. The file in this case is either a PCM file or wave file containing 16 k linear PCM audio. The 16 k linear format is used to maintain the wideband frequency range as voice data is encoded to the network codec format.

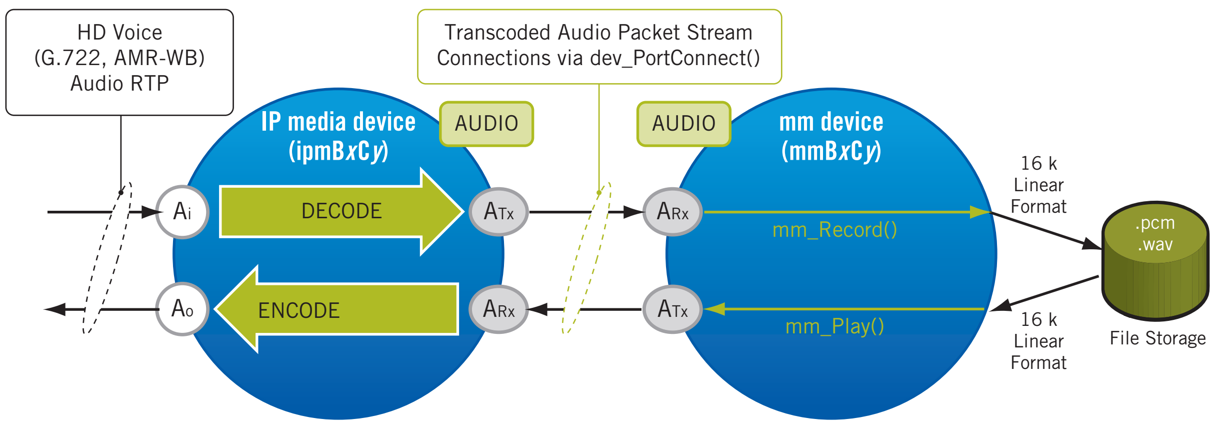

Figure 4. HD Voice Play and Record with 16 k linear Format Files

Figure 5 shows how stored HD Voice content can be played to a narrowband IP endpoint. In this scenario, the content is stored in HD Voice format with a 16 k linear PCM or wave file and down-sampled and encoded at the IPM device for transmission to a narrowband IP endpoint.

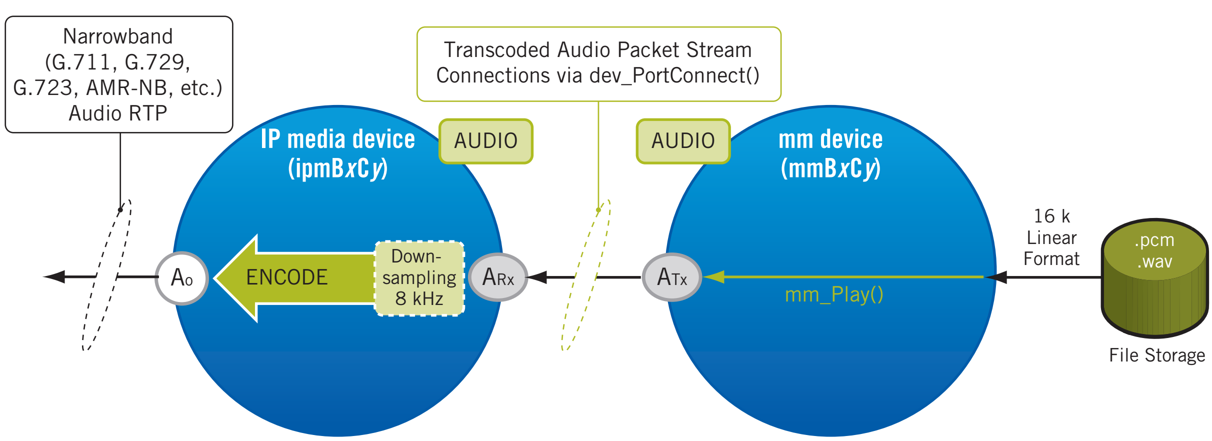

Figure 5. HD Voice Play Down-Sampled to Narrowband IP Endpoints

Figure 6 illustrates native audio play and record to an HD Voice IP endpoint. In this case, both the IPM device and the MM device are set up to stream audio natively, passing voice data from the file to the IP network without transcoding. Native audio enables higher densities and reduced latencies compared to audio transcoding. In this scenario, the Dialogic® HMP Software proprietary format G.722 or AMR-WB file is used.

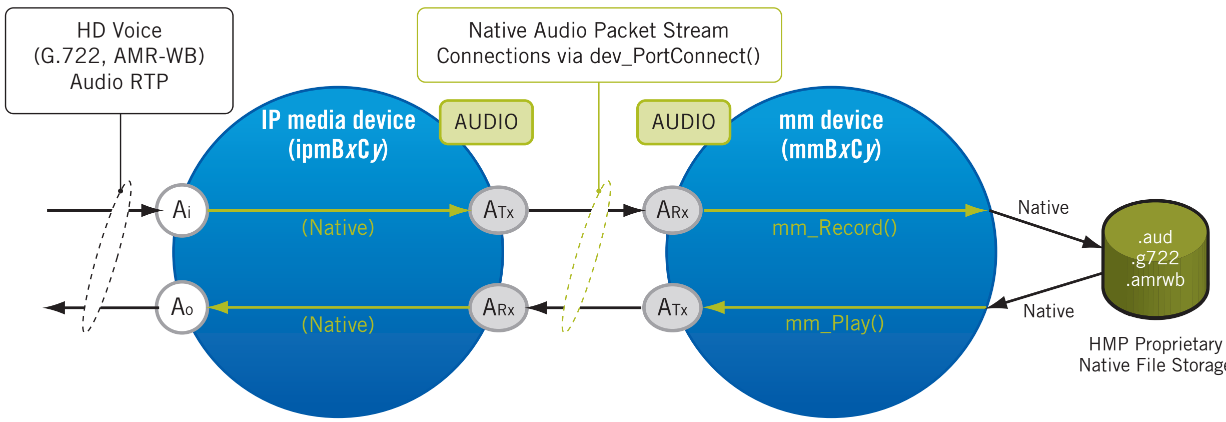

Figure 6. HD Voice Native Play and Record with Dialogic® HMP Native Files

Figure 7 highlights how multi-party distribution of stored HD Voice content can be achieved. In this scenario, one MM device is used to play HD Voice stored content simultaneously to multiple IP endpoints. The 16 k linear PCM content is encoded to the G.722 HD Voice endpoint, as well as being down-sampled and encoded to narrowband G.711 and G.729 endpoints.

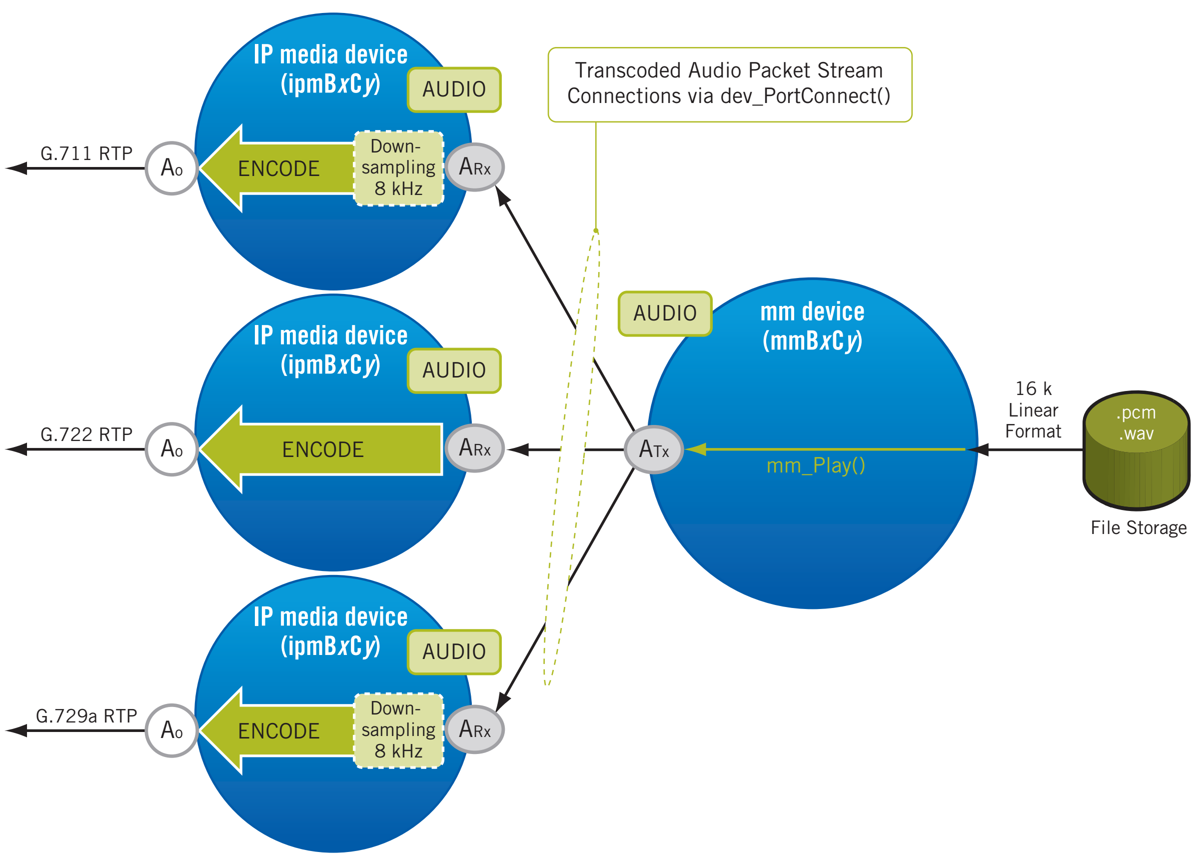

Figure 7. HD Voice Multi-Party Audio Content Distribution

Enabling HD Voice Support in Dialogic® Host Media Processing (HMP) Software Release 4.1LIN

Application Note

HD Voice Conferencing

CNF Conferencing using MCX Device

HD Voice Conferencing is supported by the Dialogic® CNF Conferencing API Library and the Multimedia (MCX) conferencing device. Only the MCX conferencing device can do audio mixing in either narrowband or HD Voice wideband format and support the proper audio fidelity range for both wideband and narrowband audio parties. The CNF device used by the Dialogic CNF Conferencing API Library and Dialogic® DCB Conferencing API Library do not support wideband mixing.

The MCX Conference device provides wideband mixed output to the HD Voice parties and narrowband mixed output to the narrowband parties. All input audio is decoded into a common linear PCM format, 16 k linear PCM for HD Voice wideband codec and 8 k linear PCM for narrowband codecs. The MCX conferencing device mixes all the HD Voice parties in 16 k linear PCM wideband format along with up-sampled narrowband parties. Similarly, the narrowband parties are mixed in 8 k linear PCM format along with the down-sampled HD Voice parties. The resulting mixed output is encoded back to the proper input audio codec format (minus the input audio). This provides the wideband frequency range output to all HD Voice parties and the narrowband frequency range output to all narrowband parties.

Figure 8 shows how HD Voice conferencing provides mixing for both wideband and narrowband parties. The scenario shows two HD Voice enabled endpoints: Party A, an HD Voice enabled IP endpoint, and Party D, an MM device for play and record in HD Voice. Also, in the scenario are two narrowband endpoints: Party B, a G.711 IP endpoint, and Party C, a G.729 IP endpoint. The MCX conference device mixes all parties and provides the combination audio with the input audio subtracted. The wideband parties get 16 k linear PCM format audio to support encoding to the appropriate wideband codec. The narrowband parties get 8 k linear PCM format audio to support encoding to the appropriate narrowband codec.

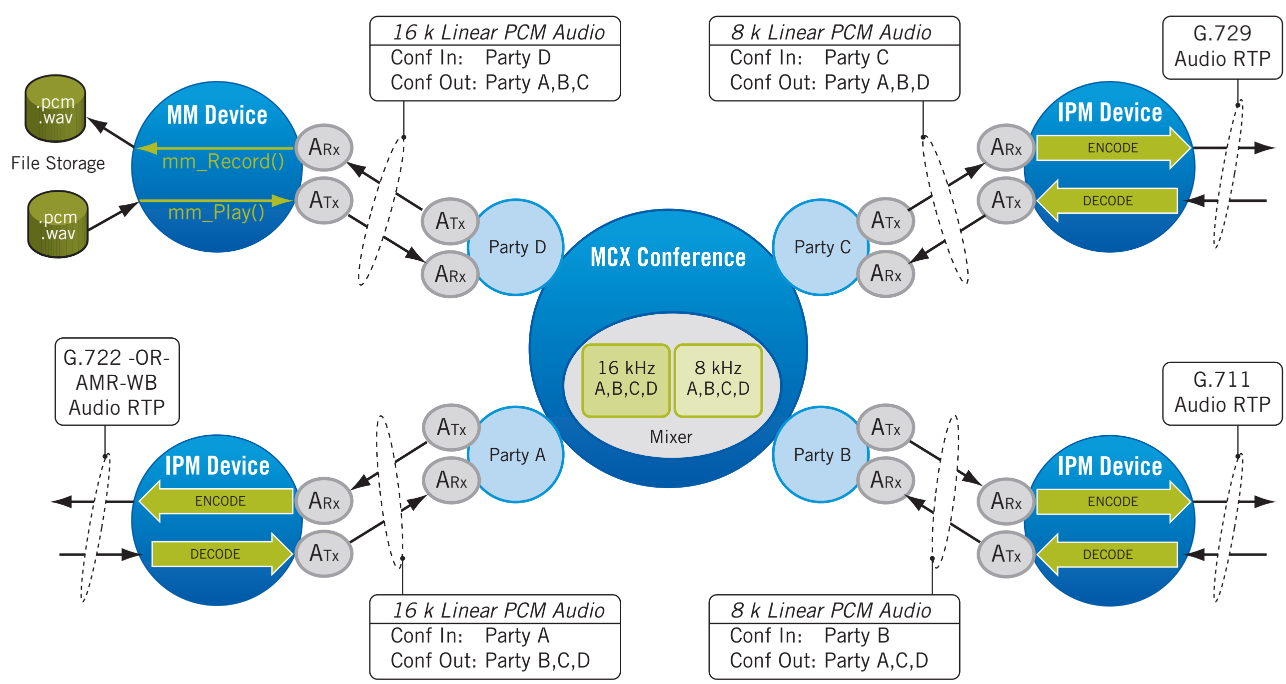

Figure 8. HD Voice Conference with Wideband and Narrowband endpoints

Note: The CNF and DCB conferencing devices do not support mixing in HD Voice. However, an endpoint using a wideband codec can still be connected to these conferences, but is down-sampled and mixed in the 8 k linear PCM narrowband format. The return audio is an up-sampled version of the mixed narrowband conference and thus does not provide HD Voice quality because the conference output frequency range is the narrowband frequency range.

HD Voice for 3G Video Calls

3G-324M (M3G) Device

In keeping with Dialogic's aim to support HD Voice for 3G Video calls, the following are targeted for future Dialogic releases:

- The Dialogic® 3G-324M API Library and Mux3G (M3G) device support for HD Voice streaming to 3G mobile video endpoints.
- The Dialogic® 3G-324M API support for AMR-WB encoding to 3G-324M Mobile devices that support negotiation of AMR-WB as part of the H.245 Terminal Capabilities Set (TCS) exchange message.
- The M3G device to support connections to all other HD Voice enabled devices and thus AMR-WB can be streamed native or transcoded to/from the 3G-324M endpoint.

The same connection rules apply for M3G devices as with other HD Voice enabled devices. The HD Voice wideband frequency range can only be maintained if the source media is supplied from an HD Voice media file or an HD Voice enabled device. Otherwise, voice data is up-sampled out to the 3G-324M AMR-WB endpoint. Likewise, AMR-WB audio from a mobile endpoint only retains its HD Voice fidelity if connected to another device that supports HD Voice. Otherwise, the AMR-WB audio is down-sampled when transported to a narrowband device.

Note: AMR-WB was added to Release 8 of the 3GPP 3G-324M specification (3GPP TS 26.111 Release 8) and might not be supported by all endpoints.

Connecting Dialogic® HMP Software HD Voice Enabled Devices

In order to support HD Voice audio wideband frequency range, the devices are connected via their packet-based ports using the Dialogic® Device Management API Library.

Dialogic® Device Management API Library

The Device Management API Library is used to make wideband connections between HD Voice enabled devices. Connections made with the dev_Connect() and dev_PortConnect() APIs are used to connect packet-based ports between devices, which can support sampling rates higher than 8 kHz. The dev_Connect() API can be used to make audio transcoded connections between devices. When audio is transcoded, it is converted from the format of the device's internal Transmit port to the format of the device's internal Receive port.

The dev_PortConnect() API is a more granular point-to-point connection method. The dev_PortConnect() API can be used to make unidirectional audio and video connections between devices using either transcoded or native connections. The dev_PortConnect() API also supports connections from one transmitter to many receivers, each independently set with transcoding enabled or disabled between TX-Rx port pairs. When audio transcoding is enabled between devices, HMP logic determines if the audio is HD Voice or requires up-sampling or down-sampling. Both Transmit and Receive devices must be configured for one of the HD Voice wideband codecs or 16 k linear PCM in order to maintain the HD Voice audio quality between endpoints.

Note: The use of xxx_listen and xxx_unlisten commands are based on 8 k Lin PCM timeslot connections. Therefore, xxx_listen and xxx_unlisten commands do not support HD Voice audio quality between devices. When xxx_listen is used to connect to an HD Voice device, the voice data is down-sampled and results in standard "toll quality" audio.

Inband DTMF Generation and Detection for HD Voice Endpoints

The Dialogic® Voice API Library and Voice (dx) device are used for inband DTMF detection and generation to HD Voice endpoints. Tone detection and generation during wideband sessions is confined to narrowband frequencies. No special custom tones in the upper frequency range are supported.

The voice device connects to an HD Voice enabled device through the standard xxx_listen() and xxx_unlisten() APIs. This means that the voice device can also be used for audio plays/record to HD Voice endpoints, but the audio will always be up-sampled from 8 k linear PCM on the play side or down-sampled to 8 k linear PCM on the record side.

Figure 9 shows how a voice device is connected to an IPM device for inband DTMF generation and detection. The connection between devices is a timeslot-based connection, so the generated DTMF tone is up-sampled before sending to the HD Voice IP endpoint. In the reverse direction, the incoming inband DTMF tone is down-sampled in order to send in a timeslot for detection at the voice device. In both cases, no wideband range frequency components are in the DTMF tones.

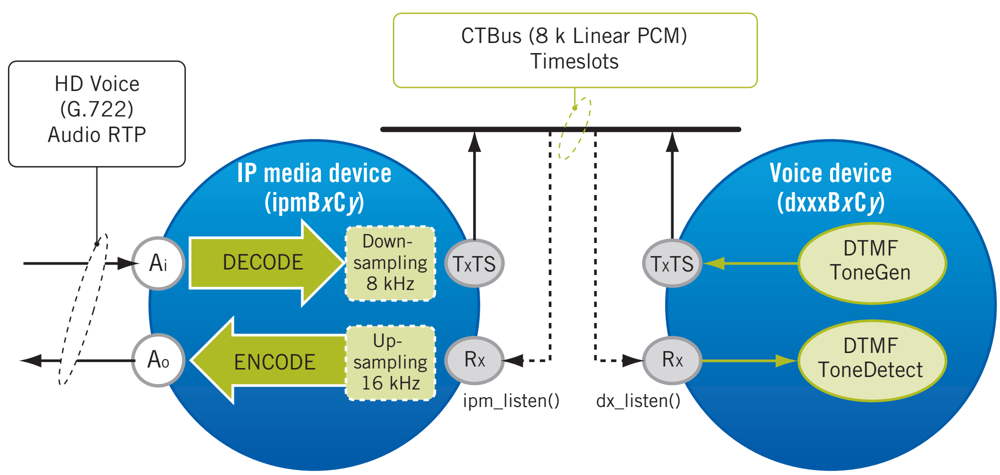

Figure 9. HD Voice Inband DTMF Generation and Detection (ipm-dx)

Note: Inband DTMF detection can be used for inband digits as well as RFC2833 digits that are re-generated into the voice timeslot. Dialogic HMP Software supports RFC2833 digit event detection for HD Voice endpoints via IPM_TELEPHONY events. See the *Dialogic® IP Media Library API Programming Guide and Library Reference* for more information.

A Note on AMR-NB and AMR-WB

Using the AMR-NB and/or AMR-WB resource in connection with a Dialogic® product described herein does not grant the right to practice the such standard(s). To seek a patent license agreement to practice the standard(s), contact the VoiceAge Corporation at http://www.voiceage.com/licensing.php.

For More Information

Are You Ready for HD Voice? Dialogic White Paper at
http://www.dialogic.com/products/docs/whitepapers/11602-hd-voice-wp.pdf

Dialogic® Host Media Processing Software Release 4.1LIN at
http://www.dialogic.com/manuals/hmp41lin/default.htm

Dialogic® HMP Software at
http://www.dialogic.com/products/ip_enabled/

Information on HD Voice at
http://www.dialogic.com/technologies/hd-voice.htm

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