VoIP GATEWAY CHIP

The Adaptive Digital VIP-P VoIP Gateway chip offers a complete silicon plus software solution designed to enable designers to increase the value of their end product thus providing a superior voice experience to the customer. Adaptive Digital's VIP-P VoIP Gateway chip provides all of the essential voice processing features needed in a VoIP gateway. The VIP-P includes Adaptive Digital's field proven algorithms including DTMF detect and conferencing. These features, in conjunction with our AT&T certified echo canceller, ensure maximum voice quality and quick time-to-market.

The VIP-P is based upon the Texas Instruments TMS320C6424 DSP, combined with Adaptive Digital's VoIP firmware and host processor API software.

1 HARDWARE FEATURES

- High-Performance
  - 400-/500-/600-/700-MHz, C64x+™Clock Rate
  - Eight 32-Bit C64x+™ Instructions/Cycle
  - 2.5-, 2-, 1.67-, 1.43-ns Instruction Cycle Time
  - 3200, 4000, 4800, 5600 MIPS
- 16-Bit Host-Port Interface (HPI)
- Enhanced Direct-Memory-Access (EDMA) Controller (64 Independent Channels)
- C64x+ L1/L2 Memory Architecture
- Two Multichannel Buffered Serial Ports
- 10/100 Mb/s Ethernet MAC (EMAC)
- Multichannel Audio Serial Port (McASP0)
- IEEE-1149.1 (JTAG™) Boundary-Scan-Compatible
- On-Chip ROM Bootloader

  Up to 111 General-Purpose I/O (GPIO) Pins
- Power
  - 3.3-V and 1.8-V I/O, 1.2-V Internal
  - 3.3-V and 1.8-V I/O, 1.05-V Internal

2 STANDARD FEATURES

- Port Count
  - VIP-24: 24 PCM, 16 LBR (Low Bit Rate Codec)
  - VIP-12: 12 PCM, 8 LBR (Low Bit Rate Codec)
- G.711 with appendices 1(Packet Loss Concealment) and 2 (silence suppression, voice activity detection (VAD), discontinuous transmission (DTX), and comfort noise generation (CNG))
- T-38 Fax Relay
- G.729 AB
- DTMF Detect
- Automatic Gain Control (AGC)
- G.168 EC Certified by AT&T Voice Quality Lab
- Tone Generate
- Tone Relay
- Supports TDM to Packet Channel
- RTP payload formatting

3 OPTIONAL FEATURES

- G.723.1A
- G.726
- G.728
- Conferencing
- Transcoding
- GSM AMR
- MELP
- Channel types: TDM to TDM, Packet to Packet, TDM Conference, Packet Conference, Conference Composite

4 DEVICE OVERVIEW

A VoIP gateway acts as a bridge between traditional telephone equipment and VoIP equipment. Traditional telephone interfaces include both analog (FXS and FXO), and digital (PCM, T1/E1 DS0). These interfaces can be found in the PSTN (Public Switched Telephone Network), PBX (Private Branch Exchange) equipment in business offices, and in residential telephone equipment. Gateways enable users of traditional telephone equipment to make use of the benefits of VoIP. Without gateways, it is impossible for VoIP equipment to place calls to users of traditional telephone equipment. Stated differently, without VoIP gateways, traditional telephone equipment and VoIP equipment could not co-exist.
By performing functions such as voice and fax compression, decompression, packetization, call routing and control signaling, a VoIP gateway enables the data infrastructure to handle voice and fax applications.

5 FUNCTIONAL DESCRIPTION

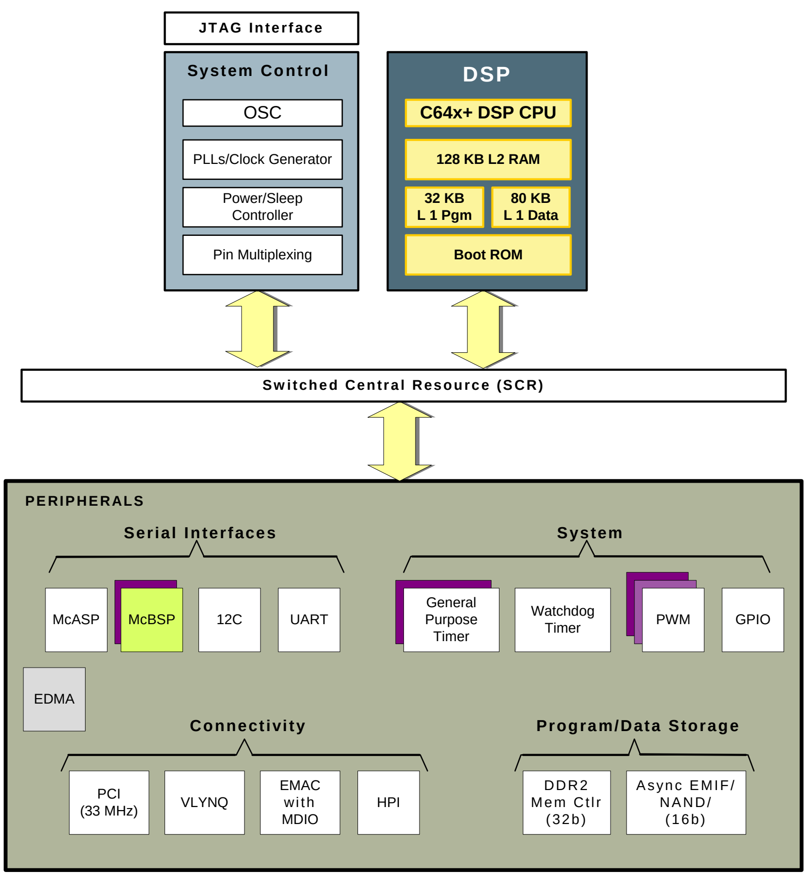

Figure 1: TMS320C6424 Functional Block Diagram

The VIP-P chip Central Processing Unit (CPU) consists of eight functional units, two register files, and two data paths. The two general-purpose register files (A and B) each contain 32 32-bit registers for a total of 64 registers. The general-purpose registers can be used for data or can be data address pointers. The data types supported include packed 8-bit data, packed 16-bit data, 32-bit data, 40-bit data, and 64-bit data. Values larger than 32 bits, such as 40-bit-long or 64-bit-long values are stored in register pairs, with the 32 LSBs of data placed in an even register and the remaining 8 or 32 MSBs in the next upper register (which is always an odd-numbered register).

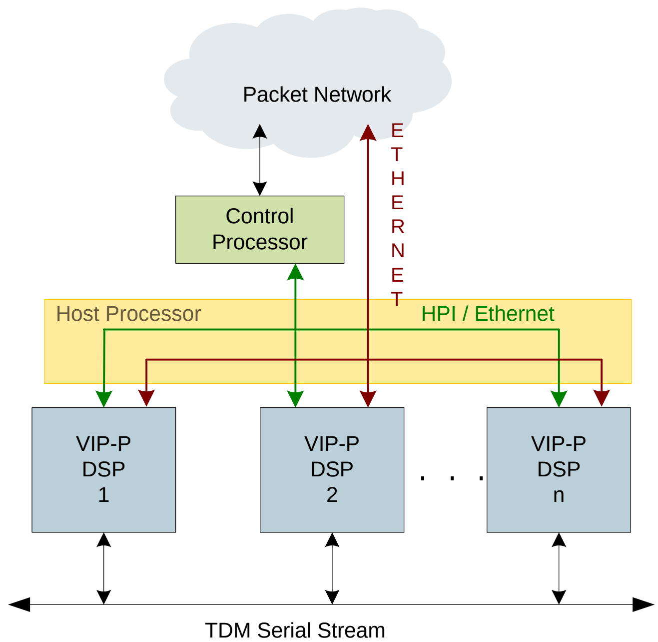

Figure 2: VIP-P VoIP Gateway system block diagram

Figure (2) shows a block diagram for a typical VoIP Gateway. A VoIP Gateway usually consists of a host control processor connected to one or more ViP-P chips. The host processor typically controls the VIP-P via either the host port interface or via Ethernet. The voice packets can be routed between the network and the VIP-P chip via the control processor. Alternatively, the VIP-P chip can be connected directly to the network via the Ethernet interface. The VIP-P chip connects to the TDM interface either via the chip's TDM serial port.

The major components in the VIP-P chip include vocoders, echo cancellation, voice quality enhancement algorithms, and telephony algorithms. The VIP-P chip supports a number of channel types: TDM to Packet, Packet to Packet, TDM to Conference, Packet to Conference, and Conference Composite. Channel setup (identification of input and output ports, vocoders, and voice algorithms), conference setup, and teardown operations are controlled by the host processor using a set of VIP-P API functions.

5.1 Channel Types

There are several types of channels: TDM to Packet, Packet to Packet, TDM to Conference, Packet to Conference, and Conference Composite.

A TDM channel is typically associated with one of the following types of telephone interfaces:

- FXO
- FXS
- T1/E1 time slot (DS0)

Each channel in a DSP is dynamically setup as any type. Frame sizes, vocoder types, and tone detection types are selected when a channel is setup.

All channels (except for the conference composite channel type) are designed to operate as full duplex channels. A full duplex channel may be configured to operate as a half duplex by setting the end points of one-half of the full duplex channel to NULL end-points.

5.1.1 TDM to Packet Channel Type

The TDM to packet channel type provides a linkage between TDM interfaces and the packet network. TDM to packet channels are used to convert PCM data, typically in G.711 or 16-bit linear format, to/from packet data. Channels of this type have two threads of processing as shown in figure (3).

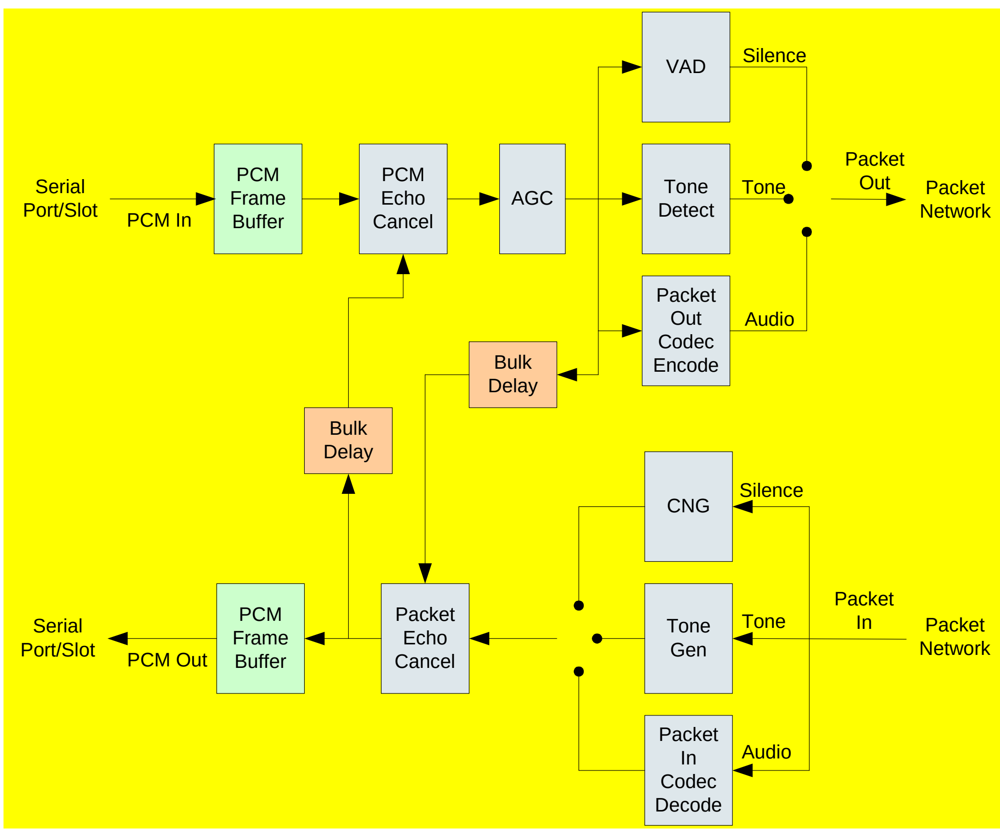

Figure 3: TDM to Packet channel

TDM to packet processing. The TDM to packet processing thread inputs PCM samples from a TDM time slot and buffers as many samples as needed to build an output frame. When enough samples are buffered, the PCM input data is optionally passed through the PCM echo canceller, automatic gain control (AGC), the tone detector (TD) and the voice activity detector.

Tone detection is optionally performed on the PCM data after the G.168 echo cancellation [1] but prior to the non-linear post-processing of the echo canceller. VIP-P notifies the host at the start and end of each tone. If tone relay is enabled, tone packets are generated for each frame until the tone ends; an end of tone packet is generated when the tone stops.

Voice activity detection is optionally performed when no tone is detected. A silence packet is generated when VAD does not detect voice activity.

Voice encoding is performed whenever tone or silence packets are NOT generated. The channel's encoder type identifies which vocoder will be used. The vocoder output is formatted and packed into the vocoder's packet payload and sent to the host for transmission. The appropriate packet (voice, tone, or VAD) is sent to the network via the packet interface.

Packet to TDM processing. The packet to TDM processing thread reads a packet from the host control processor and generates PCM samples according to the received packet type. If a silence packet is received, comfort noise is generated. If a tone packet is received, the specified tone is generated. If an audio packet is received, it is decoded according to the input packet codec type. The generated PCM samples are then optionally passed through the packet echo canceller and buffered. The buffered samples are then output to a TDM time slot.

The PCM echo canceller, packet echo canceller, AGC, VAD, tone detection, and tone relay algorithms are independent options selectable at channel setup.

5.1.2 Packet to Packet Channel Type

The Packet to packet channel type provides a linkage between packet channels. Packet to packet channels are typically used in gateway applications to perform frame size conversions, codec type conversions (transcoding), and/or echo cancellation of packet data. This type of channel can also be used to convert silence and tone packets to audio packets or to convert audio packets to silence and tone packets. Channels of this type have two threads of processing as shown in figure 4. Each VIP-P packet to packet channel is half duplex; however, these channels are always allocated in pairs (channels A and B) to allow full duplex operation.

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[1] Although not shown in the diagram, tone detection is performed using an echo canceller intermediate signal to improve the reliability of the tone detection.

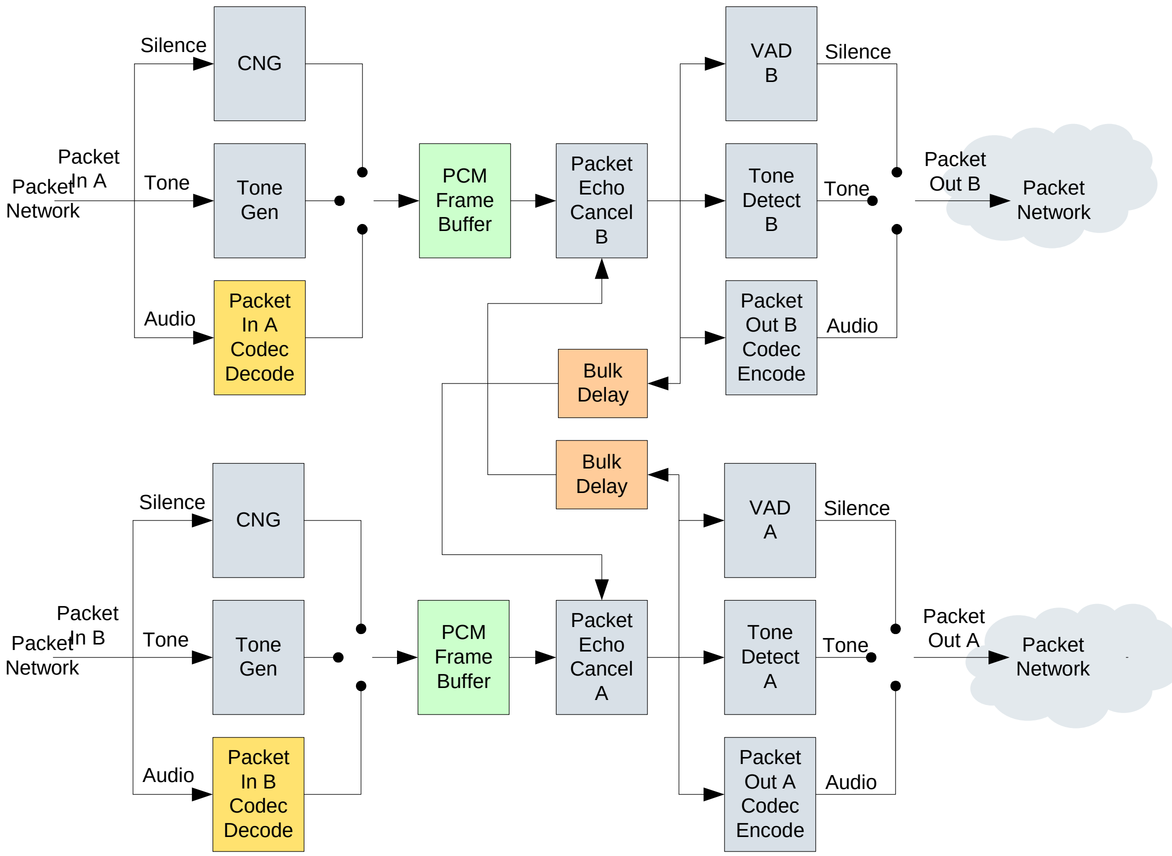

Figure 4: Packet to Packet Channel

Packet decode processing. The packet decoding thread reads a channel 's packet from the packet interface and generates and buffers PCM samples according to the received packet type. If a silence packet is received, comfort noise is generated. If a tone packet is received, the specified tone is generated. If an audio packet is received, it is decoded according to the input packet codec type. The generated PCM samples are buffered for encoding by the paired channel.

Packet encode processing. When enough of the paired channel's PCM samples are buffered to build an output frame, this PCM data is optionally passed through the packet echo canceller, the tone detector (TD) and the voice activity detector (VAD). Tone detection is optionally performed on the PCM data after the G.168 echocancellation [2]. VIP-P notifies the host at the start and end of each tone. If tone relay is enabled, tone packets are generated for each frame until the tone ends; an end of tone packet is generated when the tone stops.

Voice activity detection is optionally performed when no tone is detected. A silence packet is generated when VAD does not detect voice activity.

[2] Although not shown in the diagram, tone detection is performed an echo canceller intermediate signal to improve the reliability of the tone detection.

Voice encoding is performed whenever tone or silence packets are NOT generated. The channel's encoder type identifies which vocoder will be used. The vocoder output is formatted and packed into the vocoder's packet payload and sent to the host for transmission to the paired device.

Echo cancellation. Channel A's echo canceller cancels the echo contained in channel B's input from channel A's output. Similarly, channel B's echo canceller cancels the echo contained in channel A's input from channel B's output.

Each of the voice enhancement algorithms -packet echo canceller, VAD, and tone detection and tone relay - are independent options selectable at channel setup.

5.1.3 TDM to Conference Channel Type

The TDM to conference channel type is used to allow TDM channels to join a conference call. TDM to conference channels invoke two functions within a single processing thread as shown in Figure 5.

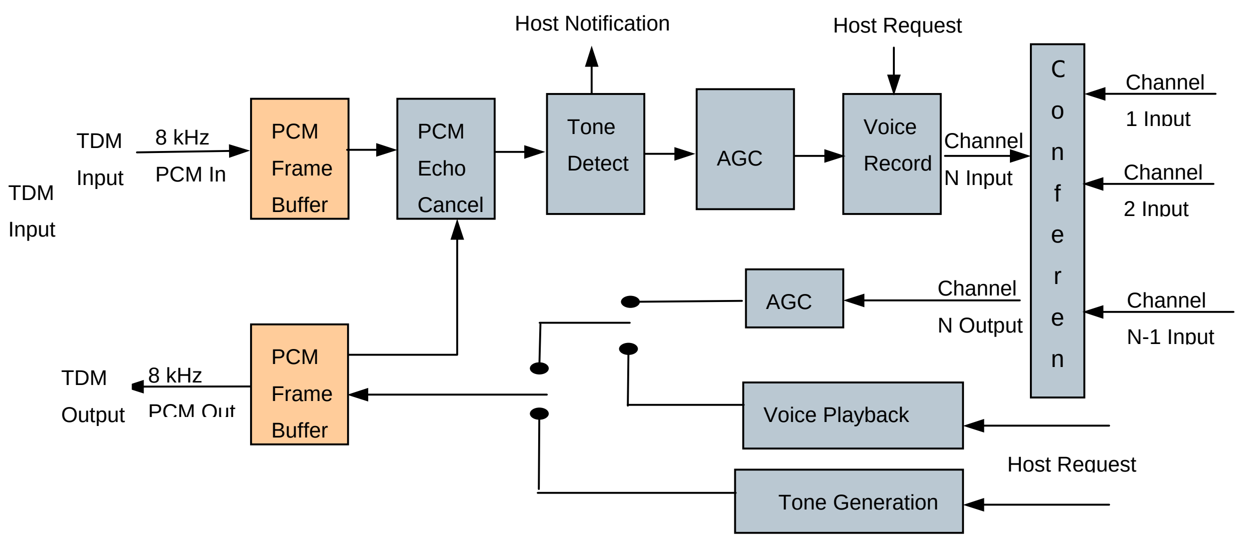

Figure 5: Conference to TDM Channel

Conference input. The conference input function inputs PCM samples from a TDM slot and buffers as many samples as needed for a frame used by the conference. When enough samples are buffered, the PCM data is optionally passed through PCM echo cancellation, tone detection, and automatic gain control (AGC) algorithms before being passed to the Conferencing algorithm for use as one of the conference inputs. The PCM data can be recorded at the host's request for future playback. The echo canceller's output is optionally sent to the DTMF detector. VIP-P informs the host of the start and end of DTMF tone bursts.

Conference output. The conference output function buffers the conference's channel-specific output data for the channel. The buffered samples are optionally passed through AGC - then output to a TDM slot. The tone generator can be used to overwrite the output signal with a host-selected tone.

The PCM echo canceller and automatic gain control algorithms are independent options selectable at channel setup.

5.1.4 Packet to Conference Channel Type

Packet to conference channel types allow packet channels to join a conferencing call. These channels invoke two functions within a single thread of processing as shown in Figure 6.

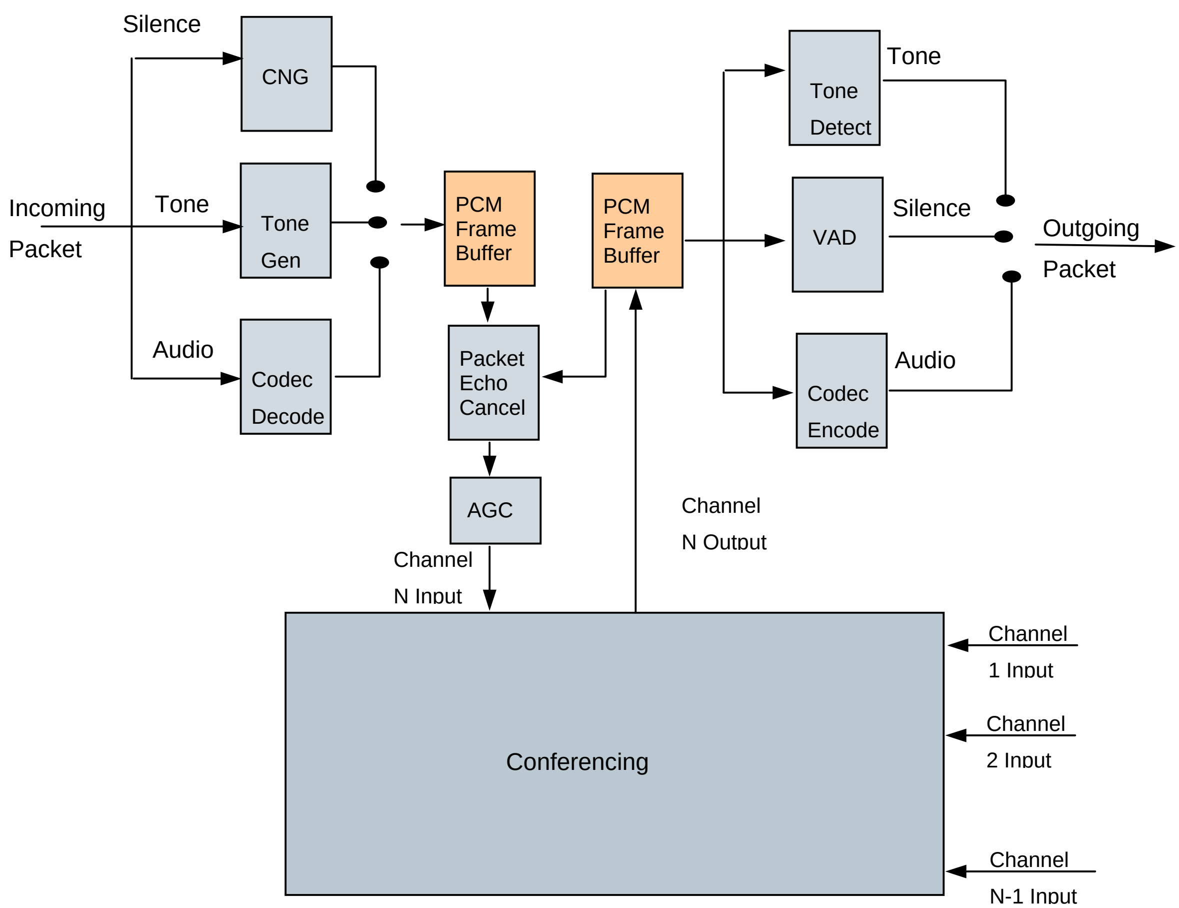

Figure 6: Conference Packet Channel

Conference input. The conference input function reads a packet from the packet interface and generates PCM samples according to the received packet type. If a silence packet is received, comfort noise is generated. If a tone packet is received, the specified tone is generated. If an audio packet is received, it is decoded according to the input packet codec type.

The generated PCM samples are then optionally passed through the packet echo canceller and automatic gain control algorithms before being passed to the Conferencing algorithm for use as one of the conference inputs.

Conference output. The conference output function buffers the conference's output data for the channel. The samples are buffered until there are enough samples to build an output frame. When enough samples are buffered, the samples are optionally passed through the tone detector (TD) and the voice activity detector (VAD). VIP-P notifies the host at the start and end of each tone. If tone relay is enabled, tone packets are generated for each frame until the tone ends; an end of tone packet is generated when the tone stops.

Voice activity detection is optionally performed when no tone is detected. A silence packet is generated when VAD does not detect voice activity,

Voice encoding is performed whenever tone or silence packets are NOT generated. The channel's encoder type identifies which vocoder will be used. The vocoder output is formatted and packed into the vocoder's packet payload and sent to the host for transmission.

The host may request that the output be overridden either with tones. This can be done using either tone relay packets or using in-band tone signaling via the voice packets. In the later case, the tone generator feeds the codec encode block with tone samples.

The packet echo canceller, tone detection, tone relay, voice activity detection, and automatic gain control algorithms are independent options selectable at channel setup. Input packets are read and output packets are generated at a rate commensurate with the conference frame size.

5.1.5 Conference Composite Channel Type

Conference Composite channel types are used to output the composite output from a conference as PCM data and/or packet data as shown in Figure 7. Conference Composite channels can also be used to report the occurrence of tones detected on the conference inputs. The differences between conference composite channels and other conference channels are:

- The conference composite channel has no input stream
- The conference composite channel is the composite of all conference members[3] signals summed together. Other conference channels remove each participant's voice before sending the conference output back each participant..
- The conference composite output can be sent to both the packet interface and to the TDM interface.

[3] The conference algorithm does not necessarily add all conference members' signals. The conference algorithm adds only a selected number of dominant speakers at any given time.

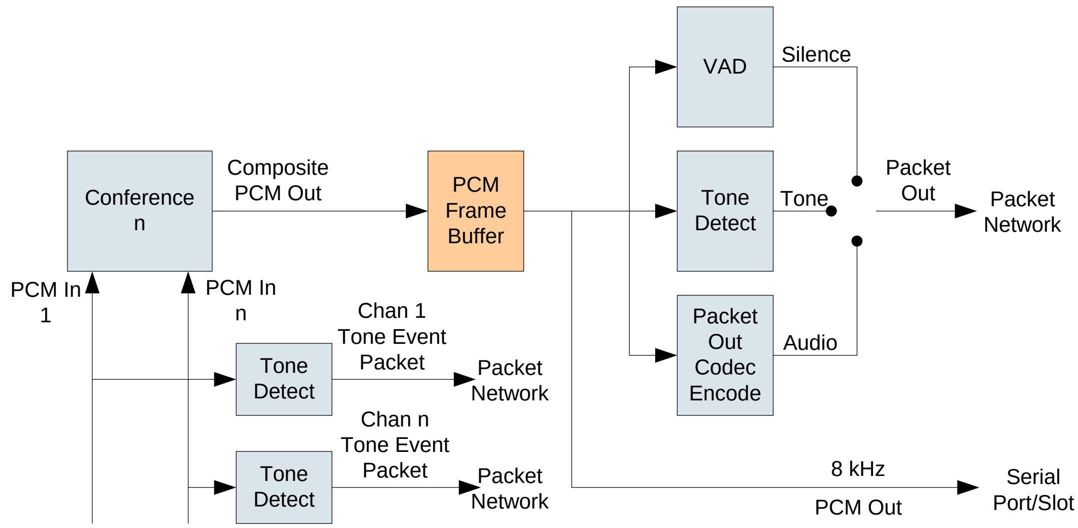

Figure 7: Conference Composite Channel

The conference composite function buffers the conference composite's output data. . The samples are buffered until there are enough samples to build an output frame. When enough samples are buffered, the samples are optionally passed through the tone detector (TD) and the voice activity detector (VAD).

VIP-P notifies the host at the start and end of each tone. If tone relay is enabled, tone packets are generated for each frame until the tone ends; an end of tone packet is generated when the tone stops.

Voice activity detection is optionally performed when no tone is detected. A silence packet is generated when VAD does not detect voice activity,

Voice encoding is optionally performed whenever tone or silence packets are NOT generated. The channel's encoder type identifies which vocoder will be used. The vocoder output is formatted and packed into the vocoder's packet payload and sent to the host for transmission.

PCM data is optionally transferred to a TDM slot.

The tone detection, tone relay, and voice activity detection algorithms are independent options selectable at channel setup. Output packets and tone event packets are generated at a rate commensurate with conference's frame size.

6 **VIP-P APIs**

The VIP-P APIs are the interface between a user's application program and VIP-P DSP cores. The APIs execute in a host control processor connected to the DSP via either Ethernet or the DSP's Host Port Interface (HPI). The APIs support multiple DSP cores/chips and use a DSP Identifier to select a particular core. The association between a DSP Identifier and a particular DSP core/chip is made by the user modified VIP-P support functions.

The APIs are provided as ANSI “C” source code. The APIs will work with any host application regardless of the operating system being used.

7 REFERENCES

1. Adaptive Digital Technologies G.PAK Users Guide

2. Texas Instruments TMS320C6424 Fixed-Point Digital Signal Processor (literature number SPRS347C)

Adaptive Digital is a strategic member of the Texas Instruments Developer Network.

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