DATA SHEET

OPEN G.PAK™

A ready to use Reference Design Kit

Adaptive Digital Technologies, Inc.

ADT G.PAK is available for the TI TMS320™ DSP Family

C64x™DSP Generation

PRODUCT DESCRIPTION

OPEN G.PAK supplies limited operation software, used to develop software and demonstrate feasibility for Voice over Internet Protocol (VoIP) applications.

OPEN G.PAK allows design engineers to build a custom digital signal processor (DSP) application in mere minutes using Texas Instruments Incorporated's Code Composer Studio™ Integrated Development Environment (IDE) in conjunction with the Windows®-based Open G.PAK configuration tool.

OPEN G.PAK supplies executables and object libraries that allow developers to easily configure, build, and test VoIP applications (digital telephones, conferencing servers, VoIP gateways, IP PBXs, etc.) that are targeted to run on Texas Instrument DSPs.

The resulting software image is downloadable into a target DSP. The customization and build process takes only a few minutes.

Built upon two previous generations of the G.PAK product; Open G.PAK brings with it the benefits of open source software.

Previous versions of G.PAK required any such customization to be done by Adaptive Digital. By incorporating Open Source into G.PAK™, Adaptive Digital has empowered its customers to take control of the DSP application by allowing them to modify the fully tested G.PAK source code themselves. This gives them a remarkable head start in their application development. By removing unneeded functionality, the DSP application fits into the most cost-effective DSP chip.

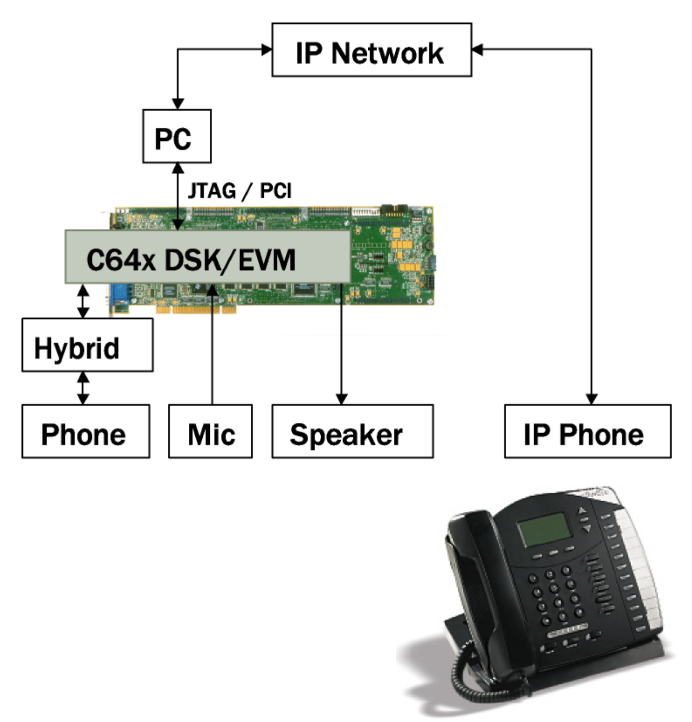
VOICE PROCESSING FUNCTIONS

G.168, AEC, Conferencing, DTMF, Tone Detect/Generate, AGC, VAD/CNG, G.711, G.728, G.729AB, G.723.1, G.726, AMR-NB, G.722.2 (AMR-WB), Noise Suppression

Simple and complete reference design.

Includes core algorithms, but allows customer to augment with additional algorithms.

Open-source allows user to differentiate end products with varying feature and density requirements.



Reference Kit Hardware

G.PAK is a scalable and configurable voice-over-packet DSP software solution that turns a digital signal processor chip into an easily controlled voice-over-packet engine. G.PAK integrates the building blocks that are required in voice-over-packet systems into a turnkey solution. System designers can therefore leverage a proven solution, allowing

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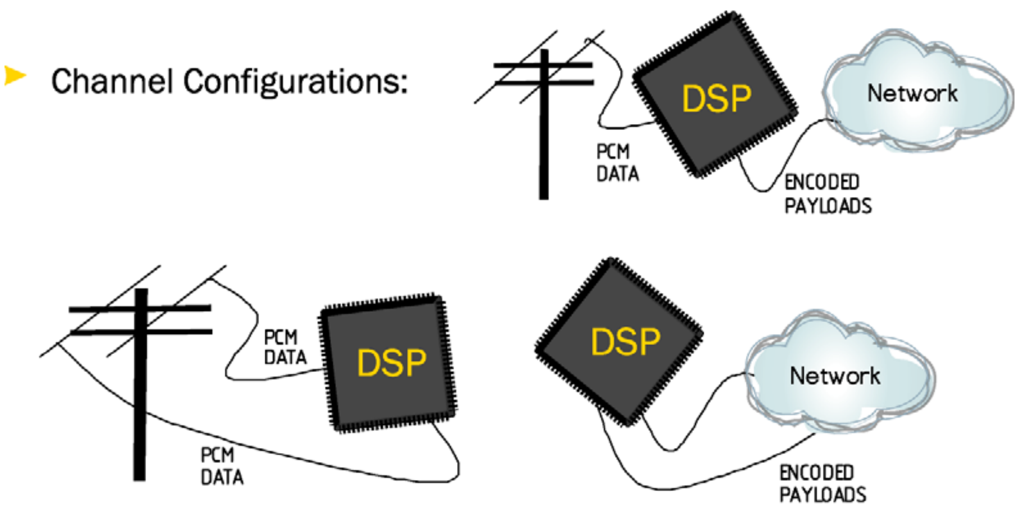
them to focus their efforts on rapid product development.

BUILD TIME CONFIGURABILITY

In order to maximize channel density, G.PAK is configured specifically for the user's application. No extra resources (MIPS and Memory) are wasted on algorithms or port configurations that are not required in the users application. Similarly, this approach allows the designer to select a more cost effective DSP solution.

CHANNEL TYPE



Each G.PAK DSP supports a fixed number of channels. The number of channels available in a DSP is dependent on the DSP type and the capabilities selected at build time.
There are several types of channels; TDM to Packet, PCM to TDM, Packet to Packet, AAL2, Circuit Data. Each channel in a DSP can be dynamically setup as any type at run time. Frame sizes, audio codec types, and tone detection types are selected when a channel is setup from capabilities configured at build time

- PCM to Packet type channels have 2 threads of processing. The first thread inputs 8 KHz samples from a serial port slot, processes, encodes, and writes packet payload data to the control processor. The second thread reads packet payload data from the control processor, decodes, processes, and outputs 8 KHz samples to a serial port slot. Each thread can use any configured frame size, codec type, and serial port slot. AGC, VAD, and Tone processing can optionally be performed on the PCM data. G.168 can optionally be used for the PCM and/or packet data. This type of channel is typically used in a voice over packet application.

- PCM to PCM type channels have 2 functions within a single thread of processing. The thread can use any configured frame size. The first function inputs 8 KHz samples from serial port slot A and outputs to serial port slot B. The second function inputs 8 KHz samples from serial port slot C and outputs to serial port slot D. G.168 can optionally be used for serial port A's data and/or serial port C's data This type of channel is typically used to provide echo cancellation and can also be used as a time slot interchanger.

- Packet to Packet type channels provide conversion of packet payload data from one type or frame size to another for a full duplex communication path. Packet payload data is read from the control processor and decoded to 8 KHz samples. The samples are then processed, encoded, and the modified packet is written to the control processor. This same process occurs for two packet paths (A to A' and B to B'). Each packet can use any configured frame size and codec type. VAD and Tone processing can optionally be performed on the decoded data. G.168 can optionally be used for the A to A' path and/or B to B' path. This type of channel is typically used in a gateway application to perform rate conversion, codec type conversion, or echo cancellation.

- Circuit Data type channels have 2 threads of processing. The first thread inputs a number of contiguous slots (8 KHz samples) from a serial port, multiplexes the samples into a single packet payload, and writes the packet payload to the control processor. The second thread reads a packet payload from the control processor, demultiplexes the samples from the payload, and outputs the samples to a number of contiguous slots on a serial port. Both threads multiplex or demultiplex the same number of samples (Mux Factor). The frame size used by both channels is dependent upon the Mux Factor. This type of channel is typically used for AAL2 Circuit Data packets.

CHANNEL DENSITY

The maximum number of channels available with G.PAK is a function of the target DSP type and the features selected at build time. Each G.PAK DSP is configured at build time to ensure that sufficient DSP memory and MIPS exist to support all channels concurrently regardless of the features selected at channel setup.

Open VoIP DSP Application Software

TIME TO MARKET

Minimize Cost

- No need to reinvent the wheel. Leverage existing application and optimized algorithms.
- By engineering at a higher level of integration, and in the "C" language, there is no need to delve into the deep detail of DSP programming.

Differentiate Your Product

- The reference design includes features that are common to many VoIP applications.
- Spend your engineering resources on features that differentiate your product.
- Differentiation is made possible by the open-source nature of the kit.

Reduce Risk

- Start with proven hardware and software to reduce development risk.
- Start with proven algorithms to reduce performance risk.
- The plug-and-play demos allow you to evaluate performance up-front.

Increase Managability

- Use a one-stop shop approach for hardware design, software, and algorithms.

Maximize Revenue

- Get your product to market faster.
- Start making sales sooner.
- Capture more of the market

Open G.PAK is:
A plug-and-play demonstration platform for VoIP
A complete open development hardware and software platform
A bundle of DSP software building blocks

Including:
Hardware reference design
Application framework in source code format
The commonly used voice algorithms
Configuration and supervision software on the PC
Development tools

Minimum Recommended Requirements

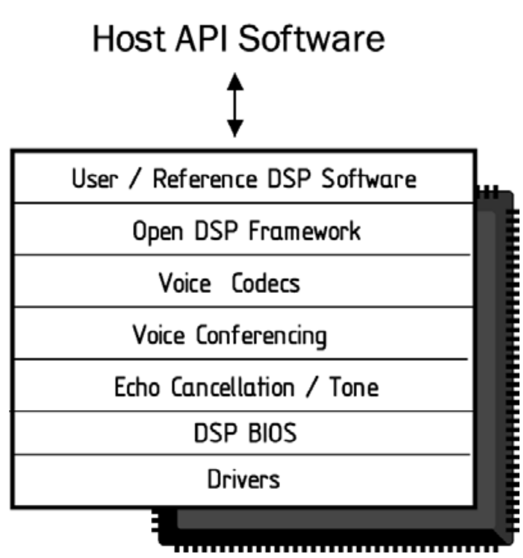
The following equipment and software are the recommend minimum requirements for using Open G.PAK:

- Texas Instrument's Code Composer Studio (Version 3.1 or later)
- Microsoft Windows_ 2000/XP
- Pentium III processor
- 500 Mb Free Disk Space
- 128 Mb Memory (RAM)
- Development board (Spectrum Digital's TMS320C6416 DSK or TMS320DM642 EVM)

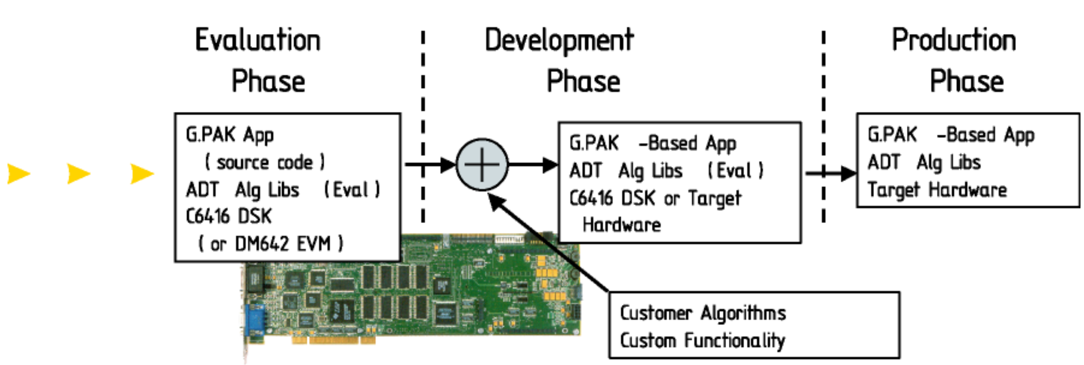
he following is recommended when interfacing Open G.PAK with a digital telephone:

- Pentium 4 – 3.0 GHz
- 1 Gb Memory (RAM

OPEN G.PAK SOFTWARE ARCHITECTURE



OPEN G.PAK DEVELOPMENT CYCLE



Sample Configurations

FULL CONFIGURATION
VoIP Loopback Demo
DSK to IP Phone Demo
Application Source
Configuration Builder
PC Supervision Software
G.711
G.726
Tone Detect/Generate
AGC
G.168
AEC
G.729AB
G.723.1
AMR

BASIC CONFIGURATION
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LOOPBACK DEMO
VoIP Loopback Demo
PC Supervision Software
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G.729AB

IP PHONE DEMO
VoIP Loopback Demo
DSK to IP Phone Demo
IP Phone
G.711
G.729AB

BUSINESS MODELS

- Algorithm: Customer develops DSP application using Adaptive Digital's algorithm libraries.
- G.PAK: Adaptive Digital delivers entire DSP software (custom built) as downloadable binary image, with host API software.
- Open G.PAK: Adaptive Digital delivers DSP software with framework source code, CCS project, and algorithm libraries.

CUSTOM DESIGN SERVICES

- Custom design at fixed cost
- Participation in design review
- Consulting services

CONTACT INFORMATION

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