XMOS AVB Design Guide

Version 4v0 - Draft

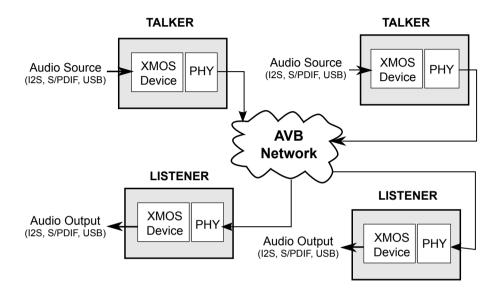


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1 Summary

The XMOS Audio Visual Bridging (AVB) reference design can be used to stream synchronized audio over an ethernet network. The XMOS solution is based on event-driven programmable devices, which can transmit and receive multiple audio streams, and implement both talker and listener functionality.



XMOS AVB Features

- Supports all endpoint requirements: ethernet interface, packet processing, timing synchronisation, configuration/stream setup and media rate recovery can all be handled on device.
- Supports emerging AVB ethernet standards such as *IEEE 802.1 as*, *IEEE 801.1 Qav* and *IEEE P1722*.
- Flexible software based design implements both hardware interfaces and protocol layers in the same environment allowing, flexibility in system design and easy modification.
- Multiple audio hardware interfaces supported such as I2S, TDM and S/PDIF.

2 XMOS AVB Specification

Functionality				
Provides ethernet interface, audio transport, precise timing protocol clock synchronise and media clock recovery to streamed audio over ethernet.				
Support	ed Standards			
Ethernet IEEE 802.3 (via MII)				
AVB QoS	IEEE 802.1 Qav			
Precise Timing Protocol	IEEE 1588v2 or IEEE 802.1as			
AVB Audio Over Ethernet	IEEE 1722			
Audio Streaming	IEC 61883-6			
Suppo	rted Devices			
XMOS Devices	XS1-G4			
	XS1-G2			
	XS1-L2			
Req	uirements			
Development Tools	XMOS Desktop Tools v9.9.1 or			
	later			
Ethernet	$1 imes MII$ compatible $100 ext{Mbit PHY}$			
or				
2 imes MII compatible 100Mbit PHY				
or				
1 $ imes$ GMII compatible Gigabit PHY + programmable logic device				
Audio input/output device (e.g. ADC/DAC audio CODEC)				
	PLL/Frequency synthesizer chip to generate CODEC system clock			
Boot/Storage	Compatible SPI Flash Device			
Licensing and Support				
Reference code provided under license from XMOS.				
Contact support@xmos.com for details.				
Reference code is maintained by X	MOS Limited.			

3 Ethernet AVB Standards

Ethernet AVB consists of a collection of different standards that together allow audio and video to be streamed over ethernet. The standards allow synchronized, uninterrupted streaming with multiple talkers and listeners on a switched network infrastructure.

3.1 802.1as

802.1as defines a precise timing protocol based on the *IEEE 1558v2* protocol. It allows every device connected to the network to share a common global clock. The protocol allows devices to have a synchronized view of this clock to within microseconds of each other, aiding media stream clock recovery and coordinated AVB traffic control.

This protocol is implemented in the XMOS AVB solution.

3.2 802.10av

802.1Qav defines a standard for buffering and forwarding of traffic through the network using particular flow control algorithms. It uses the global clock provided by 802.1as to synchronize traffic forwarding and gives predictable latency control on media streams going through the network.

The XMOS AVB solution implements the requirements for endpoints defined by 802.1Qav. This is done by traffic flow control in the transmit arbiter of the ethernet MAC component.

3.3 802.1Qat

802.1Qat defines a stream reservation protocol that provides end-to-end reservation of bandwidth across an AVB network.

This protocol is not currently implemented in the AVB solution. It could be implemented as part of the user application on an XMOS device or by a separate host processor communicating with an XMOS device.

3.4 Emerging standards

3.4.1 IEEE P1722

IEEE P1722 defines an encapsulation protocol to transport audio streams over ethernet. It is complementary to the AVB standards and in particular allows timestamping

of a stream based on the 802.1as global clock.

The XMOS AVB solution handles both transmission and receipt of audio streams using *IEEE P1722*. In addition it can use the *802.1as* timestamps to accurately recover the sample rate clock of the audio to match on the listener side.

3.4.2 IEC 61883-6

IEC 61883-6 defines an audio data format that is contained in IEEE P1722 streams.

The XMOS AVB solution uses IEC 61883-6 to convey audio sample streams.

4 XS1 Architecture

The XS1 architecture consists of one or more processing cores, called XCores™, which have the following properties:

- Each XCore can execute up to eight threads concurrently, each at a speed of up to 100 MIPS. Each thread has a dedicated register set enabling it to operate as a logical core.
- The eight threads share a single 64 KByte unified memory with no access collisions.
- Integer and fixed point operations are provided for efficient DSP and cryptographic operations.
- I/O general purpose pins are provided, which can be programmed from software. Thread execution is deterministic and hence each thread can implement a hard real-time I/O task, regardless of the behavior of other threads.
- I/O pins are grouped into logical ports of width 1, 4, 8, 16 and 32 bits. Each port incorporates serialization/deserialization, synchronization with the external interface and precision timing.
- Each XCore incorporates eight timers that measure time relative to a 100 MHz reference clock.

The architecture is designed to support standard programming languages such as C. Extensions to standard languages, libraries, or the use of assembly language provide access to the full benefits of the instruction set.

4.1 XMOS XS1 Programmable Devices

The XS1 family is the first available implementation of the XS1 architecture. It includes the XS1-G4 device that integrates four XCore processors. Each device is connected to a high performance switch interconnect via four internal links called XMOS Links™. Each link is capable of transferring data at 800 Mbits/second. The switch provides full connectivity between the cores on the programmable device, and also provides up to sixteen external XMOS Links. Each external link is capable of transferring data at up to 400 Mbits/second.

Other members of the family include the XS1-L1, XS1-G2 and XS1-L2. XS1 devices can be used standalone, or can be connected together using XMOS Links to create a network of XCore processors.

4.2 Software Libraries

The XS1 architecture implements hardware as software, so that hardware solutions are just software libraries. These software solutions are compiled and linked like ordinary C code into a binary file which can be loaded and executed on the device.

The XMOS AVB Reference Software is a library of source files that can be included in a larger application (templates and demos are provided with the reference design). The application can then be compiled and deployed onto a device to provide your AVB hardware solution.

4.3 XC Language

The XMOS originated XC language [1] is based on C. XC is a concurrent and real-time programming language designed to target the XS1 architecture. XC programs are easy to write and debug, leading to programs that are free from deadlocks, race conditions and memory violations and can be compiled to produce high performance multicore designs.

XC uses channels to provide high-speed bidirectional communication and synchronization between channel ends within threads on the same processor, a different processor on the same chip or a different chip.

5 XMOS AVB System Description

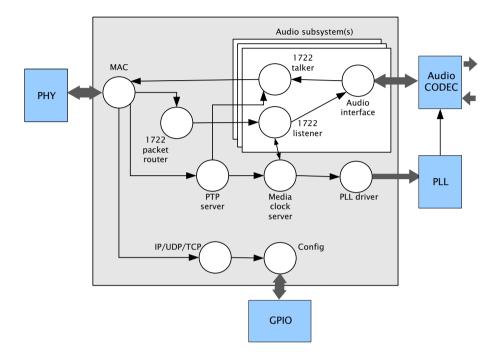


Figure 1: The structure of an XMOS AVB endpoint

Figure 1 shows the overall structure of an XMOS AVB endpoint. An endpoint consists of five main components:

- The ethernet MAC
- The precise timing engine (PTP)
- Audio streaming components
- The media clock server
- Configuration and other application components

5.1 Ethernet MAC Component

The MAC component provides ethernet connectivity to the AVB solution. To use the component, a physical interface must be attached to the XCore ports to provide either an MII interface (for 100Mbps etherenet) or GMII interface (for Gigabit ethernet).

The MAC component supports two features that are necessary to implement AVB standards with precise timing and quality constraints.

- Timestamping allows receipt and transmission of ethernet frames to be timestamped with respect to a clock (i.e a 100MHz reference clock can provide a resolution of 10ns).
- Bandwidth control allows different channels to have different priorities and bandwidth restrictions to allow steady flow of outgoing media stream packets. The implementation provides flow control to satisfy the requirements of an AVB endpoint as specified in the upcoming 802.1Qav standard.

5.1.1 Single Port 100MBit interface

The single port 100MBit component consists of five threads (each running at 50MIPs or more) that must be run on the same core. These threads handle both the receiving and transmission of ethernet frames. The MAC component can be linked (via channels) to other components/threads in the system. Each link can set a filter to control which packets are conveyed to it via that channel.

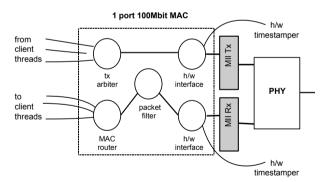


Figure 2: Single 100MBit/s MAC component thread diagram

All configuration of the channel is managed by a client C API, which configures and registers the filters. Details of the API used to configure MAC channels can be found in the header files in the src/mac/client/ directory within the reference design software.

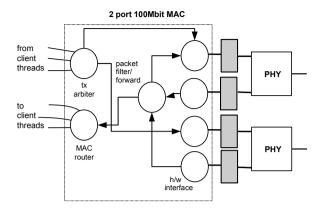


Figure 3: Dual 100MBit/s MAC component thread diagram

5.1.2 Dual Port 100MBit interface

The dual port 100MBit component consists of seven threads. These threads handle the receiving, filtering, forwarding and transmission of ethernet frames. Traffic coming in on one port is forwarded to the other port unless it is specifically addressed to the mac address of the endpoint or a broadcast or multicast packet.

The client API for the dual port MAC is the same as the single port MAC. Each receive of a frame informs the application which source port the frame came from. Similarly each transmit needs to be directed to a port (or broadcast to both).

5.1.3 Single port Gigabit interface

The gigabit interface communicates with an external programmable logic chip which interfaces to the GMII signal coming from a gigabit PHY. For details on the IP for this device and the hardware setup, please contact support@xmos.com.

The single port gigabit interface consists of four threads which must be run at 100 MIPS (i.e. for an XS1-G4 they must be the only threads on one core). In addition, two other threads are required on another core for buffering transmission of packets. The gigabit MAC filters out AVB audio packets and packets destined for the endpoint MAC address. These packets are then buffered in a 40k packet buffer which is routed to the rest of the chip at 300MBit/s.

5.2 Precise Timing Protocol Component

The precise timing protocol component (PTP) provides a system with a notion of global time on a network. The component supports the *IEEE 1588v2* timing protocol and the upcoming *AVB 802.1as* timing protocol. It allows synchronization of the presentation and playback rate of media streams across a network (see Figure 4).

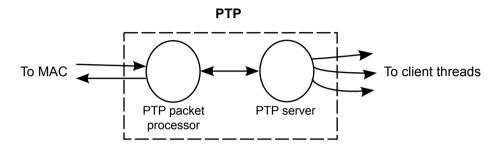


Figure 4: PTP component thread diagram

The timing component, which consists of two threads, connects to the ethernet MAC component and provides channel ends for clients to query for timing information. The component interprets PTP packets from the MAC and maintains a notion of global time. The maintainance of global time requires no application interaction with the component.

The PTP component can be configured at runtime to be a *PTP grandmaster* or a *PTP slave*. If the component is configured as a grandmaster, it supplies a clock source to the network. If the network has several grandmasters, the potential grandmasters negotiate between themselves to select a single grandmaster. Once a single grandmaster is selected, all units on the network synchronize a global time from this source and the other grandmasters stop providing timing information. Depending on the intermediate network, this synchronization can be to sub-microsecond level resolution.

Client threads connect to the timing component via channels. The relationship between the local reference counter and global time is maintained across this channel, allowing a client to timestamp with a local timer very accurately and then convert it to global time, giving highly accurate global timestamps.

Client threads can communicate with the server using the API described in the file:

src/ptp/client/ptp_client.h.

• The PTP system in the endpoint is self-configuring, it runs automatically and gives each endpoint an accurate notion of a global clock.

• The global clock is *not* the same as the sample rate clock used to time sampling (though it can be used to create the sample clock)

5.3 Audio Components

5.3.1 AVB Streams, Channels, Talkers and Listeners

Audio is transported in streams of data, where each stream may have multiple channels. Endpoints producing the streams are called *Talkers* and those receiving them are called *Listeners*. Each stream on the network has a unique 64-bit stream ID.

Routing is done using layer 2 ethernet addresses. Each stream is sent from a particular source MAC address to a particular destination MAC address. The destination MAC address may be a multicast address (i.e. several Listeners may receive it). In addition, AVB switches can reserve an end-to-end path with guaranteed bandwidth for a stream. This is done by the Talker endpoint advertising the stream to the switches and the Listener registering to receive it. If sufficient bandwidth is not available, this registration may fail.

Streams carry their own *presentation time* (the time that samples are due to be output) allowing multiple Listeners that receive the same stream to output in sync.

- Streams are encoded using the 1722 AVB transport protocol.
- All channels in a stream must be synchronized to the same sample clock.
- All the channels in a stream must come from the same Talker.
- Routing of audio streams uses ethernet layer 2 routing, which can be either unicast (one-to-one) or multicast (one-to-many).
- Routing is done at the stream level. All channels within a stream must be routed to the same place. However, a stream can be multicast to several Listeners, each of which picks out different channels.
- A single end point can be both a Talker and Listener.
- The stream ID is the only information you can obtain about a stream before registering to listen to it. Any other information about the stream must be communicated by a higher level protocol (see Section 5.5).

5.3.2 Internal Routing

As described in the previous section, an IEEE P1722 audio stream may consist of many channels. These channels need to be routed to particular audio I/Os on the

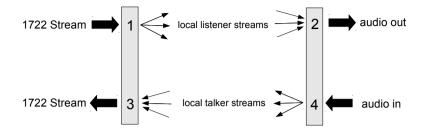


Figure 5: Internal routing via local streams

endpoint. To achieve maximum flexibility the XMOS design uses intermediate *local* audio streams to route audio.

Figure 5 shows the breakdown of 1722 streams into local streams. The Figure shows four points where transitions to and from local streams occur.

- 1. When a 1722 stream is received, its channels are mapped to local listener streams. The simplest (and most efficient) mapping is for one 1722 stream to map to one local stream. This mapping can be configured dynamically (can be changed at runtime by the configuration component).
- 2. The digital hardware interface maps local listener streams to audio outputs. This mapping is fixed and is configured statically in the software.
- 3. Several local talker streams can be combined into a 1722 stream. This mapping is dynamic.
- 4. The digital hardware interface maps digital audio inputs to local talker streams. This mapping is fixed (i.e. cannot be changed at runtime).

The configuration of the mappings is handled through the 1722 client API. This can be found in the file src/avb_1722/client/avb_1722.h.

5.3.3 Audio Talker

The talker component consists of one thread which creates *IEEE P1722* packets and passes the audio samples onto the MAC. Audio samples are passed to this component via a shared memory sample FIFO representing a local talker stream. Samples are pushed into this fifo from a different thread implementing the audio hardware interface. The packetizer thread removes the samples and combines them into *IEEE P1722* ethernet packets to be transmitted via the MAC component.

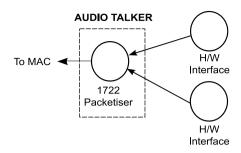


Figure 6: Audio talker thread diagram

When the packets are created the timestamps are converted to the time domain of the global clock provided by the PTP component, and a fixed offset is added to the timestamps to provide the *presentation time* of the samples (*i.e* the time at which the sample should be played by a listener).

The audio talker can be configured on how to combine local audio streams to become *IEEE P1722* streams. However, since samples are passed via a shared memory interface a talker can only combine talker streams that are created on the same core as the talker.

5.3.4 The 1722 Packet Router

The *IEEE P1722* packet router component routes *IEEE P1722* audio packets to the listener components in the system. It controls the routing by stream ID. The router is configured over a shared memory interface using the same client functions that control the listener components. It is important that any configuration threads that affect the packet router are on the same core as the packet router.

5.3.5 Audio Listener

The audio listener component takes *IEEE P1722* packets from the packet router and converts them into a sample stream to be fed into a buffer thread. Each audio listener component can listen to several *IEEE P1722* streams. The configuration of which streams it listens to and how they are split into local listener streams is configured by a channel interface from a separate configuration thread.

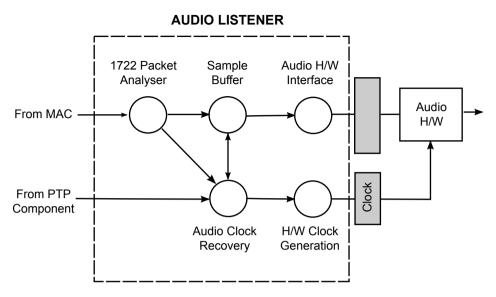


Figure 7: Listener thread diagram (h/w assisted audio clock recovery)

5.3.6 Audio Hardware Interfaces

The audio hardware interface that the audio component connects to can be configured to specific hardware.

- A hardware interface component may have to option to connect to a listener component or a talker component.
- An interface may have a link with the media clock server to drive its timing.
- The mapping between inputs/outputs and local audio streams is usually fixed and can be configured by a parameter to the function implementing the audio h/w thread. See the specific interface header file for details.

5.4 Media Clocks

A media clock controls the rate at which information is passed to an external media playing device. For example, an audio sample clock that governs the rate at which samples should be passed to an audio codec. An XMOS AVB endpoint can keep track of several media clocks.

A media clock can be synchronized to one of three sources:

- An incoming clock signal on a port.
- The rate of the incoming samples (provided by the *IEEE P1722* timestamps) on an incoming *IEEE P1722* audio stream.
- The global PTP clock.

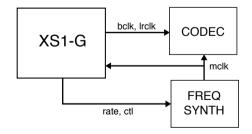
A hardware interface can be tied to a particular media clock, allowing the media output from the XMOS device to be synchronized with other devices on the network.

All media clocks are maintained by the media clock server component. This component takes one thread of processing and maintains the current state of all the media clocks in the system. It then periodically updates other components with clock change information to keep the system synchronized. The set of media clocks is determined by an array passed to the server at startup.

The media clock server component also receives information from the audio listener component to track timing information of incoming *IEEE P1722* streams. It then sends control information back to ensure the listening component honors the presentation time of the incoming stream.

5.4.1 Driving an external clock generator

A high quality, low jitter master clock is required to drive an audio codec and this clock must be synchronized with an AVB media clock. The XS1 chip cannot provide this clock directly but can provide a lower frequency source for a frequency synthesizer chip or external PLL chip. The frequency synthesizer chip must be able to generate a high frequency clock based on a lower frequency signal (e.g a chip such as the Cirrus Logic CS2300 or similar). The recommended configuration is as in the block diagram below:



The XS1 device provides control to the frequency synthesizer (which is often a divided down clock to drive the higher required rate) and the frequency synthesizer provides the master clock to the codec. The sample bit clocks and word clocks are then provided to the codec by the XS1 device.

To drive an external clock generator requires one thread connected to the media clock server. See src/audio/media_clocks/external for details.

5.5 Configuration and application threads

The AVB standard protocols control the time synchronization, streaming and routing of audio data. However, it is likely that a higher level configuration protocol will be required to configure an AVB endpoint.

Such a protocol needs two parts: discovery and control. The discovery part must determine higher level information about other endpoints on the network. For example:

- Discover which other Talkers/Listeners are on the network
- Discover which streams are available and meta-information about them (sample rate, description, global clock synchronization participation *etc.*)

The control part controls the device. For example it controls:

- The streams the Talker outputs/Listener inputs.
- Audio input/output (sample rate, gain, dsp etc.)
- Other non-audio aspects.

The AVB software solution includes a port of the uIP protocol stack which is a small memory footprint stack that can be used for UDP/TCP communication to aid implementation of an upper layer configuration protocol. Future software updates will include an implementation of Zeroconf standards (IP link local addressing, multicast DNS and DNS service discovery) to aid in initialisation and discovery. These libraries will aid in supporting configuration protocols such as the forth-coming 1722.1 AVB standard.

6 System/Application Design

6.1 XMOS AVB Endpoint Capability

An XMOS AVB endpoint provides the ability to take IEEE 1722 audio streams from ethernet and output the audio data.

The number of audio channels the device can handle depends on the XMOS device used. The XS1 platform is very flexible and can provide other functions alongside audio (depending on how much audio is used) - this includes DSP functionality, controlling inputs and displays on a device or controlling non-AVB ethernet communication.

The amount of audio available for the XS1-G4 and XS1-G2 devices is detailed in the following sections. See Section 10 for more information on chip resource usage and how these figures were determined. Please note that in a final application the exact capability depends on the type of digital audio interface, the mapping between 1722 and local streams, the complexity of the routing etc. and these figures are meant only as a rough guide.

- The maximum channel count figures assume that more than two channels are used per AVB stream to maximize channel count.
- The maximum channel count assumes I²S or similar (e.g you cannot get maximum number of channels if all are S/PDIF).
- At 100MBit/s, the capability on the XS1 for higher bit-rates are bounded by bandwidth and buffering in line with the AVB standard.
- For very high channel counts, it is assumed that a multi-channel multiplexed 1-wire protocol (e.g. TDM) is used to reduce the required pin count.

6.1.1 XS1-G4 Device - 100Mbit/s

Sample Rate	AVB Streams	Audio channels
48Khz	9 in/9 out	32 in/32 out
96Khz	6 in/6 out	16 in/16 out

6.1.2 XS1-G2 Device - 100Mbit/s

Sample Rate	AVB Streams	Audio channels
48Khz	4 in/4 out	8 in/8 out
96Khz	2 in/2 out	4 in/4 out

6.1.3 XS1-G4 Device - Gigabit

Sample Rate	AVB Streams	Audio channels
48Khz	20 in/20 out	72 in/72 out
96Khz	10 in/10 out	36 in/36 out

6.1.4 2 × XS1-G4 Device - Gigabit

Sample Rate	AVB Streams	Audio channels
48Khz	40 in/40 out	120 in/120 out
96Khz	20 in/20 out	60 in/60 out

6.2 Board Design

The key PCB components for an XMOS AVB solution are:

- 1. XS1-G4 or XS1-G2 chip (or $2 \times XS1-G4$)
- 2. 100 Mbit/s MII ethernet phy (100 Mbit) or Gigabit GMII ethernet phy + CPLD (Gigabit)
- 3. SPI flash for boot image loading and persistent storage
- 4. PLL/frequency synthesizer chip
- 5. An Audio codec or processor

To aid board design please refer to the schematics that can be found at:

www.xmos.com/support/silicon

This page includes reference designs for XMOS chips (including BOM costs). The following schematics show examples connecting to additional hardware:

Schematic	Component
XC-2	100 Mbit Etherenet PHY, SPI Flash
XC-3	2 imes 100 Mbit Ethernet PHY
XC-2G	Gigabit Ethernet PHY, CPLD (for GMII)
Audio Interface Board	Frequency synthesizer chip, Audio Codec

It is highly recommend to add an XSYS connector for an XTAG2 device onto a board for debugging.

6.3 Software Configuration/Design

Adapting the reference software to a particular application involves:

- 1. Adding specific functionality (configuration, dsp etc.)
- 2. Creating a top-level main function to run the required components.
- 3. Adapting compile time configuration parameters in avb_conf.h.

One of the most important aspects of the specific functionality is the configuration layer (see Section 5.5).

The top-level main determines which components run on which cores and their configuration (including the port mapping to specific hardware). The best way to create a top-level is to copy and adapt one of the demos. Instructions on how to create your own application files and build them can be found in the README in the src directory of the reference software.

6.4 Firmware upgrading

Every part of the XMOS AVB solution is implemented in software. This gives great flexibility for field upgrades. Firmware can be upgraded over ethernet. A forthcoming library from XMOS will help provide secure firmware upgrades storing boot images on a standard SPI flash chip. These boot images can be transferred to the device via ethernet over an application specific protocol.

7 Licensing and Support Information

7.1 Reference Software License

The XMOS AVB Reference Software is distributed under license from XMOS. For information about licensing please contact XMOS.

7.2 Support

The reference design is maintained by XMOS and is provided "as is" under the terms described in Section 7.1. For details of priority support agreements and pricing please contact XMOS. Any bugs and/or enhancement requests may be reported with a support ticket at www.xmos.com/support.

8 XMOS AVB Software Reference

8.1 Directory Structure

The AVB Reference Software is organised into a directory hierarchy described below:

```
AVB Reference Software
_bin . . . . . Directory for binaries
build......Build directory
 _avb......AVB component build files
 _demo......Demo build files
audio ...... Audio handling code
   _ codecs . . . . . . . . . . . Hardware specific audio codec configuration code
   _ interfaces.......Digital audio interface code
   _local_audio_streams......Local audio stream management code
   media_clocks ...... Media clock management code
  avb_1722.....IEEE P1722 audio streaming code
   _client......Component API
   _ server..... Component implementation
  demo.......Demo application code
  mac......MAC component code
   _client ...... Component API
  __ server...... Component implementation
  misc...... Misc. AVB code (for debugging)
  ptp......Precise timing protocol component code
   _client......Component API
   _uip.....ulP IP/UDP/TCP component code
```

8.2 Useful Reference Files

File and Description
README
Reference software README file
doc/release_notes
Reference software release notes
doc/CHANGELOG
Reference software change log
src/avb_conf_defaults.h
File showing the defines that can be overridden in avb_conf.h.
src/demo/xc-2/audio_if/*
A good simple example showing a talker/listener demo interfacing audio I ² S.
build/demo/xc-2/Makefile
Makefile for the above example.
src/mac/client/ethernet_*_client.h
MAC component API
<pre>src/ptp/client/ptp_client.h</pre>
PTP component API
src/avb_1722/client/avb_1722_client.h
Audio component API
src/avb_1722/client/xtcp_client.h
IP/TCP/UDP component API

9 Bandwidth Requirements

The AVB standard requires audio data to be split into packets to be transmitted over ethernet along with meta-information specified by the IEEE 1722 transport protocol. This meta-information incurs an overhead on the bandwidth of each stream of data.

The protocol overhead is detailed in the following table:

Protocol	O verhead (bytes)
Interframe gap	20
Ethernet header	18
IEEE 1722 header	24
61883-6 audio header	8
CRC	4

Each stream of audio data can contain several multiplexed channels of audio. The higher the number of channels per stream, the more efficient the audio transport is in terms of bandwidth. Note that the IEC 61883-6 standard recommends transmitting single channel streams as stereo with the right channel blank.

After the header in each packet, audio data is stored in AM824 format which pads 24-bit data to 4 byte quadlets. The frame rate is 8KHz so a 48KHz stream has 6 samples per packet, a 96Khz has 12 samples per packet and so on.

The following table shows the bandwidth for streams at different sample rates with different number of channels per stream.

S ample Rate	Channels/Stream	Mbps
48KHz	1	7.81
48KHz	2	7.81
48KHz	4	10.88
48KHz	8	17.02
48KHz	16	29.31
48KHz	32	53.89
96KHz	1	10.88
96KHz	2	10.88
96KHz	4	17.02
96KHz	8	29.31
96KHz	16	53.89
192KHz	1	17.02
192KHz	2	17.02
192KHz	4	29.31
192KHz	8	53.89

Note that the higher the number of channels per stream, the better the bandwidth usage (e.g. 4×8 channels streams uses less bandwidth than 16×2 channel streams). The IEEE 1722 standard also specifies that only 75% of available bandwidth can be used for AVB traffic.

10 Chip Resource Usage

10.1 Available Chip Resources

Each XMOS device has a set of resources detailed in the following table. The resource are split amongst different cores on the device which may affect how resources can be used:

D evice	MIPS	M emory	P orts
XS1-G4	1600	256k	64×1bit, 16×4bit,
			8×16 bit, 4×32 bit
XS1-G2	800	128k	32×1 bit, 8×4bit,
			4×16 bit, 2×32 bit

The following sections detail the resource required for each component. Please note that the memory requirements for code size should be taken as a rough guide since exact memory usage depends on the integration of components (which components are on which core etc.) in the final build of the application.

10.2 Ethernet Component

Each endpoint requires a drive for the ethernet phy and the precise protocol timing component. The resource usage for these is:

Component	MIPS	Memory	P orts
100Mbit/s Ethernet	250	~15k + buffering	6×1 bit + 2×4 bit
100Mbit/s 2-Port Ethernet	350	~15k + buffering	6×1 bit + 2×4 bit
Gigabit Ethernet	500	~80k	6×1 bit + 2×4 bit

10.3 PTP Component

Every AVB endpoint must include a PTP component.

Component	MIPS	M emory	P orts
PTP	100	~37k	None

10.4 Media Clock Component

Every AVB endpoint must include a media clock server.

Component	MIPS	M emory	P orts
Media Clock Server	50	~1k	None

If the endpoint drives an external PLL then a PLL driver component is required.

Component	MIPS	M emory	P orts
PLL driver	50	~0.5k	1×1-bit +
			config ports.

10.5 Audio Component(s)

Each endpoint may have several listener and talker components. Each listener/talker component is capable of handling 8 *IEEE P1722* streams and up to 24 channels of audio.

Component	MIPS	M emory
1722 processing listener	100	~5k
1722 processing talker	50	~5k

If the endpoint needs to process more than 4 *IEEE 1722* streams or has more than one listener component then the endpoint requires a 1722 packet router component.

Component	MIPS	M emory
1722 packet router	100	~5k

The amount of resource required for audio processing depends on the interface and the number of audio channels required. The overheads for the interface are:

Component	MIPS	M emory	Ports
I^2S	see below	~0.5k	3×1 bit +
			1×1bit per data in/out
TDM/I8S	see below	~0.5k	3×1 bit
			1×1bit per 8 data in/out
S/PDIF	see below	~2k	1×1bit per data in/out

In addition the following resources are required depending on the number of channels. The following table shows how much resource is required depending on the numbers of channels and the interface.

Interface	S ample Rate	Audio Channels	MIPS
I ² S	48Khz	up to 16 in/16 out	50
I ² S	96Khz	up to 8 in/8 out	50
TDM/I8S	48Khz	up to 16 in/16 out	50
S/PDIF	48Khz	2 in	100
S/PDIF	96Khz	2 in	100
S/PDIF	48Khz	2 out	50
S/PDIF	96Khz	2 out	50

Note that the several instances of the audio interface component can be made (*e.g.* you could use 100 MIPS to handle 24 channels of I²S). The following table shows how much buffering memory is required depending on the number of audio channels.

Sample Rate	A udio Channels	M emory
48Khz	n in/m out	0.5k × (n+m)
96Khz	n in/m out	$1k \times (n+m)$

10.6 Configuration Component

The uIP IP/UDP/TCP stack requires the following resource.

Component	MIPS	Memory
uIP server	50	~30k

In addition there are the application dependant threads controlling other I/O etc. For general configuration and slow I/O a minimum of 1 thread (50 MIPS) should be reserved.

Bibliography

[1] Douglas Watt and Richard Osborne and David May. XC Reference Manual (8.7). Website, 2008. http://www.xmos.com/published/xc87.

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