# Meeting ResE Requirements: A Simulation Study

Felix Feifei Feng Feng.fei@samsung.com

Geoffrey M. Garner gmgarner@comcast.net

SAMSUNG Electronics

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# Outline

□ ResE A/V application performance requirements

□ Simulation results and conclusions

- Delay variation with multiple CBR streams
- Delay under overload conditions
- Delay without source segmentation

□ Summary of conclusions

# **Delay Requirement of Applications**

#### □ ITU-T Y.1541

Network Performance Parameter	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5
IP packet transfer delay (IPTD)	100 ms	400 ms	100 ms	400 ms	1 s	U*

□ ITU-T Recommendation G.114 recommends the following general limits for one-way transmission time (assuming echo control already taken care of):

- 0 to 150 ms: preferred range (<30ms, user does not notice any delay at all, <100ms, user does not notice delay if echo cancellation is provided and there are no distortions on the link)</li>
- 150 to 400 ms: acceptable range (but with increasing degradation)
- above 400 ms: unacceptable range

#### □ Apportionment of end-to-end requirement to ResE

- End-to-end applies to a 22500 km reference path
  - E.g., G.114 indicates propagation delay 5  $\mu s/km$  due only to propagation in fiber (6  $\mu s/km$  for submarine cable systems
- Rules are given for allocating performance to portions of the path
- Rules for allocating a portion to the residence are TBD; we assume that the allocation to the residence will be less than 10 ms

# Delay Requirement of ResE Applications (Cont.)

- Most stringent delay requirement is for high-fidelity audio applications (see reference [1])
- □End-to-end audio transmission delay should not exceed 10 15 ms
  - Some of the applications involve multiple traverses of ResE network
  - Consideration of the multiple traverses, delays in the intermediate and end audio equipment on the order of 1 – 2 ms, and possible 6 ms air delay from a speaker, indicates that it is desirable that ResE transport end-toend delay not exceed 2 ms

Also consider that home sized network will have maximum of 7 Ethernet hops (1 traversal of ResE)

End-to-end delay requirement for ResE: < 2 ms for up to 7 hops

# Jitter/Wander Requirements of Applications

see reference [3]

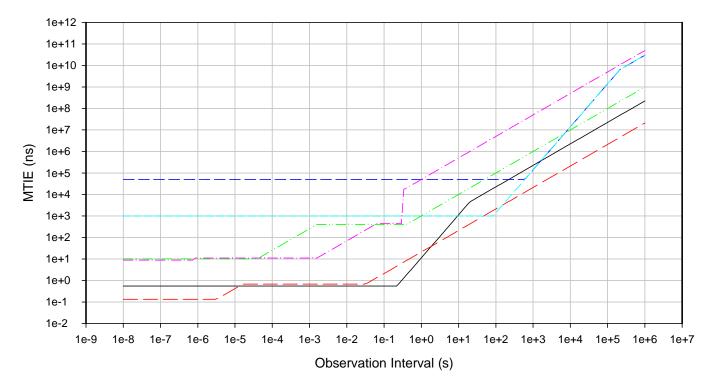
Requirement	Uncompresse d SDTV	Uncompressed HDTV	MPEG-2, with network transport	MPEG-2, no network transport	Digital audio, consumer interface	Digital audio, professional interface
Wide-band jitter (Ulpp)	0.2	1.0	50 μs peak-to-peak phase	1000 ns peak-to-peak phase	0.25	0.25
Wide-band jitter meas filt (Hz)	10	10	variation requirement (no measurement	variation requirement (no measurement filter specified)	200	8000
High-band jitter (Ulpp)	0.2	0.2	filter specified)		0.2	No requirement
High-band jitter meas filt (kHz)	1	100			400 (approx)	No requirement
Frequency offset (ppm)	±2.79365 (NTSC) ±0.225549 (PAL)	±10	±30	±30	±50 (Level 1) ±1000 (Level 2)	±1 (Grade 1) ±10 (Grade 2)
Frequency drift rate (ppm/s)	0.027937 (NTSC) 0.0225549 (PAL)	No requirement	0.000278	0.000278	No requirement	No requirement

### Network Interface MTIE Masks of Applications

see reference [3]

	Uncompressed SDTV (SDI signal)
— — —	Uncompressed HDTV (SDI signal)
	MPEG-2, after netwk transport (Ref. Pts. D and E)
	MPEG-2, no netwk transport (Ref. Pts. B and C)
	Digital Audio, Consumer Interfaces (S/P-DIF)
	Digital Audio, Professional Interfaces (AES3)

#### Network Interface MTIE Masks for Digital Video and Audio Signals



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# Jitter/Wander Requirements for ResE Synch

- Must enable A/V applications to meet end-to-end jitter and wander requirements (MTIE masks on previous slide)
- □Note that ResE gets only a budget allocation
  - High fidelity audio applications may make multiple traverses of ResE
  - •Video may be delivered through one or more service provider networks [3]
- □Note that there is also jitter and wander due to granularity of application time stamps relative to ResE synchronization signal
- Therefore, ResE synchronization jitter/wander should be well within the MTIE masks
- Assume that, in the future, ResE will carry uncompressed digital video
- Then the interface jitter requirements for a synchronization signal delivered to any ResE node are
  - •Wide-band jitter  $\leq$  fraction of 0.5 ns, measured with 10 Hz high-pass filter
  - •High-band jitter  $\leq$  fraction of 0.1 ns, measured with 100 kHz high-pass filter

# Jitter/Wander Requirements for ResE Synch (Cont.)

□The interface wander for a synchronization signal delivered to any ResE node must be somewhat<sup>(Note 1)</sup> below the following MTIE mask

$$MTIE(S) \le \begin{cases} 0.5 \text{ ns} & 0.0318 \text{ s} \le S \le 0.213 \text{ s} \\ 11S^2 \text{ ns} & 0.213 \text{ s} \le S \le 1.909 \text{ s} \\ 21S \text{ ns} & S \ge 1.909 \text{ s} \end{cases}$$

□This mask is the lower envelope of the application MTIE masks (with some rounding of the level values)

Shortest observation interval (0.0318 s) corresponds to 10 Hz [3]

□(Note 1) :The precise amounts by which they are below the lower envelope depend on the actual budget allocations for ResE and the jitter/wander added due to application time stamp granularity

# Summary of ResE A/V Requirements

Guaranteed QoS attributes over a small diameter (home-sized) network with a maximum of 7 Ethernet hops

- Guaranteed bandwidth, once a stream is established
- Latency less than 2 ms
- Packets are not dropped (PLR < 10<sup>-8</sup>)

Time synchronization supplied to ResE nodes where applications are mapped and demapped having low jitter and approaching zero wander

- •Wide-band jitter  $\leq$  fraction of 0.5 ns, measured with 10 Hz high-pass filter
- •High-band jitter  $\leq$  fraction of 0.1 ns, measured with 10 Hz high-pass filter
- Wander must be somewhat below the following MTIE mask

 $MTIE(S) \le \begin{cases} 0.5 \text{ ns} & 0.0318 \text{ s} \le S \le 0.213 \text{ s} \\ 11S^2 \text{ ns} & 0.213 \text{ s} \le S \le 1.909 \text{ s} \\ 21S \text{ ns} & S \ge 1.909 \text{ s} \end{cases}$ 

#### Can current Ethernet meet the A/V requirements?

□Simulation results for the transport of multiple time-sensitive traffic streams over current Ethernet

Baseline assumptions:

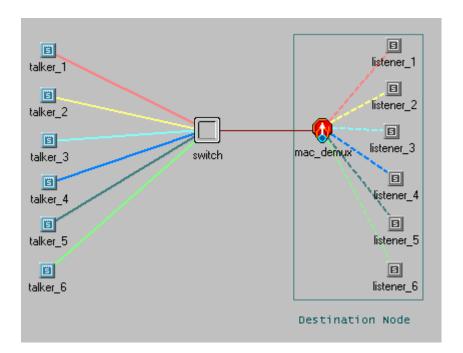
- Time sensitive traffic has constant bit rate, and has high priority
- Best-effort traffic has low priority
- Timing for a time-sensitive traffic stream is recovered at the network egress via filtering (e.g., using Phase-Locked-Loop (PLL))
  - •The requirements for these filters should be practical in consumer electronics.
  - We assume a 2nd order, linear filter with 20 dB/decade roll-off, 1 Hz bandwidth and 0.1 dB gain peaking.
  - •Note: The alternative approach using a free-running clock at the egress instead of PLL filtering is not considered, because it must buffer enough data to prevent buffer underflow or overflow for the duration of the audio or video application. For  $\pm 100$ ppm clocks and video or audio applications on the order of hours, this would imply buffering some number of seconds worth of data. This amount of delay would be added to the application end-to-end delay

# Case 1, Background

□In a network where each CBR source (time-sensitive traffic) sends packets based on their own free-running clocks, it is possible that all these sources have a same nominal rate but with small offsets.

 The streams will beat against each other, which results in undesirable delay variation at the egress

# Case 1, Network Assumptions



Utilization of the link between switch and mac\_demux: 54%

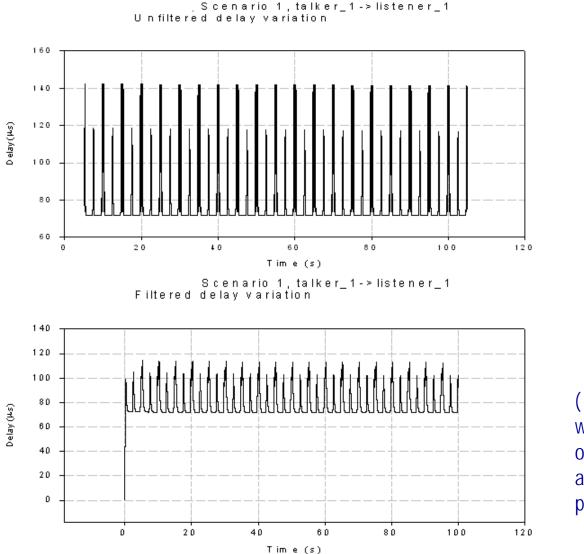
□CBR sources (talker\_1~talker\_6) have the same nominal rate of 4000pkts/s, but differ slightly (within a frequency tolerance). Packet size is a constant, equal to 2048 bits (see Reference [2]).

□All traffic turned on at 5s.

CBR source	talker_1	talker_2	talker_3	talker_4	talker_5	talker_6
Offset	0	-100ppm	+100ppm	0	-50ppm	+50ppm

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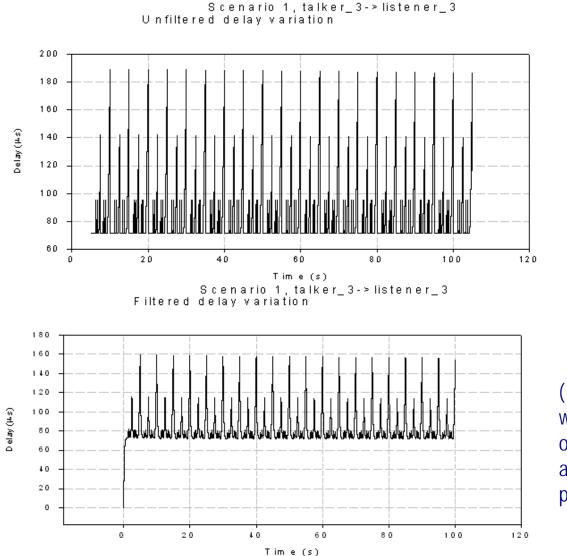
#### Case 1, Unfiltered and Filtered Delay Variation



(2nd order, linear filter with 20 dB/decade rolloff, 1 Hz bandwidth and 0.1 dB gain peaking)

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#### Case 1, Unfiltered and Filtered Delay Variation (Cont.)



(2nd order, linear filter with 20 dB/decade rolloff, 1 Hz bandwidth and 0.1 dB gain peaking)

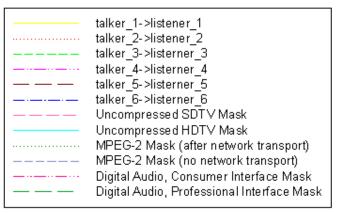
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#### Case 1, MTIE Curve

and comparison with video and audio masks 1e+7 1e++6 1e+5 1e+4 MTIE (ns) 1e+3 1e+2 1e+1 1e+0 1e-1 0.0001 0.001 0.01 0.1 10 100 1000 1 Observation Interval (s)

Scenario 1 stream MTIE curves



# Case 1, Conclusions

□For competing CBR traffic streams whose rates differ slightly from nominal rate, timing recovery for the time sensitive traffic streams by filtering the streams at the egress (e.g., with a PLL function) will not enable the respective jitter and wander requirements to be met (the requirements are embodied in the MTIE masks).

□Note that the case here is not worst-case

- Time-sensitive streams with larger packet sizes would give worse performance (larger delay variation)
- Networks with more time-sensitive streams and/or higher link utilization due to the time-sensitive steams (e.g., 70%) will give worse performance (larger delay variation)

An alternative scheme that transports timing through some other means is needed: Timing synchronization

# Case 2, Background

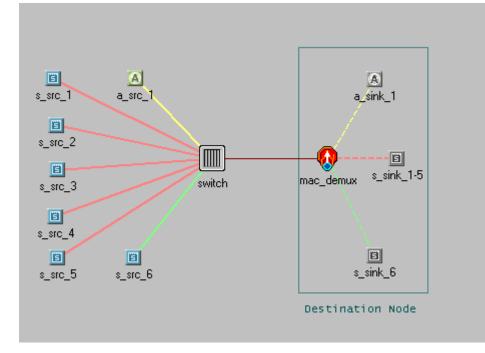
#### Current Ethernet doesn't have admission control function.

Network can be overloaded

#### Backward pressure flow control is not suitable

- Most time-sensitive stream applications follow a "guaranteed or no service" model.
- It needs additional processing and buffering in source nodes, which are cost-sensitive consumer electronics

### Case 2, Network Assumptions

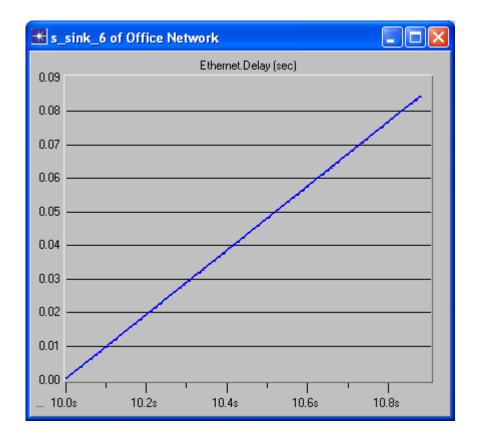


Utilization of the link between switch and mac\_demux: >100%

■Best effort source a\_src\_1 has a self similar arrival (1000pkts/s, H=0.7). Packet size has a uniform distribution between 1Kbits and 12Kbits.

□CBR sources (s\_src\_1~s\_src\_6) have the same nominal rate of 8000pkts/s. Packet size is a constant, equal to 2Kbits.

□All traffic turned on at 10s.



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# Case 2, Conclusions

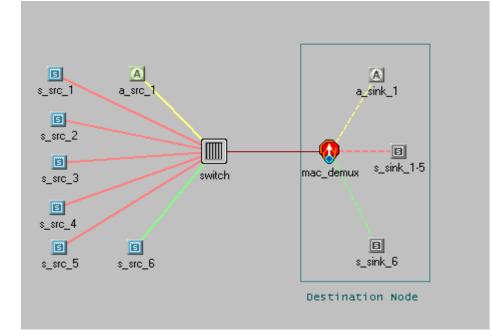
- □Without admission control, network can be overloaded and the network performance is deteriorated.
- Deterministic low latency and low jitter for data delivery can be provided only if the availability of network resources is guaranteed and intermediate bridges are appropriately configured along the entire transmission path.
- □A subscription protocol for explicit negotiation (admission control) of network resources and configuration of bridges is required
  - Such a subscription protocol provides the function of establishing end-toend streams in the layer 2 Residential Ethernet.
    - •The subscription protocol could be further interfaced with the signaling protocol of upper layer applications.

# Case 3, Background

Additional to the baseline assumption, we further assume:

- A certain timing synchronization mechanism (e.g. a IEEE1588 like scheme), is employed in Ethernet
- Admission control is employed in Ethernet. Average bandwidth of timesensitive traffic streams will not overload the network.
- In case that there are large period time-sensitive traffic streams with their average bandwidth satisfying the admission control criteria, these streams may still cause large bunching delays

### Case 3 - Scenario 1, Network Assumptions

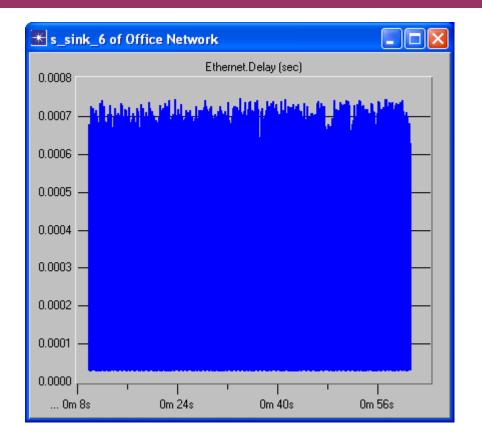


Utilization of the link between switch and mac\_demux: 47%

■Best effort source a\_src\_1 has a self similar arrival (1000pkts/s, H=0.7). Packet size has a uniform distribution between 1Kbits and 12Kbits.

- □CBR sources s\_src\_1~s\_src\_5 have the same nominal rate of 500pkts/s. Packet size is a constant, equal to 12000 bits.
- □CBR sources s\_src\_6 has the nominal rate of 8000pkts/s. Packet size is a constant, equal to 1000bits.
- □All traffic turned on at 10s.

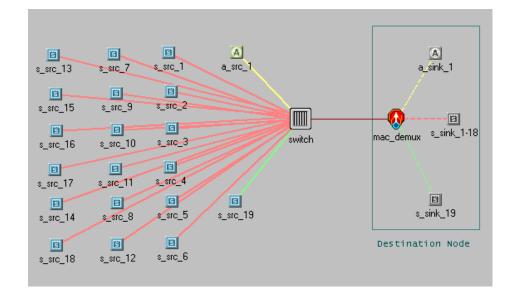
### Case 3 - Scenario 1, Delay



#### □Note that this case is not worst-case

 CBR streams experiencing more contending time-sensitive streams (either in one hop or in several hops) could have worse delay performance.

### Case 3 - Scenario 2, Network Assumptions

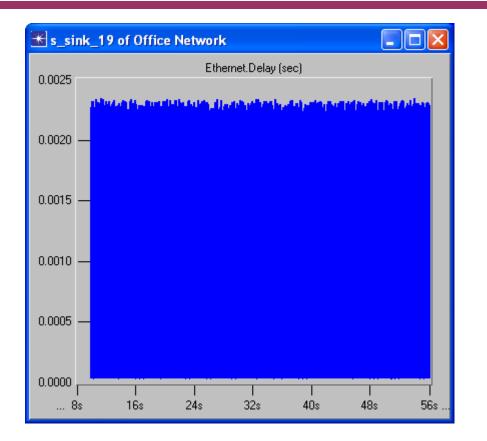


Utilization of the link between switch and mac\_demux: 87%

■Best effort source a\_src\_1 has a self similar arrival (1000pkts/s, H=0.7). Packet size has a uniform distribution between 1Kbits and 12Kbits.

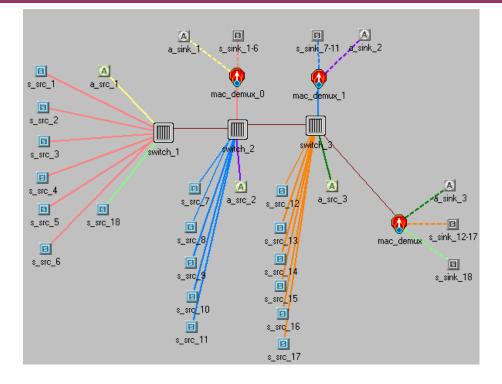
- □CBR sources s\_src\_1~s\_src\_18 have the same nominal rate of 320pkts/s. Packet size is a constant, equal to 12000 bits.
- □CBR sources s\_src\_19 has the nominal rate of 8000pkts/s. Packet size is a constant, equal to 1000bits.
- □All traffic turned on at 10s.

### Case 3 - Scenario 2, Delay



#### Delay results in this scenario somewhat exceed the 2ms requirement

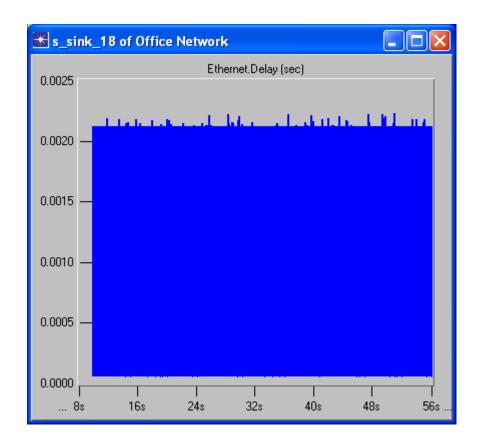
# Case 3, Scenario 3, Multi-hop Example



Link Utilization Switch1->Switch2: 53% Switch2->Switch3: 65% Switch3->mac\_demux: 40%

- □s\_src\_1~6: 500pkts/s, packet size 12Kbits
- □s\_src\_7~11, 800pkts/s, packet size 12Kbits
- □s\_src\_12~17, 320pkts/s, packet size 12Kbits
- □s\_src\_18, 8Kpts/s, packet size 1Kbits
- □a\_src\_1~3, 1Kpts/s, self-similar arrival (H=0.7), packet size has a uniform distribution between 1Kbits and 12Kbits.
- □ All traffic turned on at 10s.

### Case 3, Scenario 3, Delay



Delay results in this scenario somewhat exceed the 2ms requirement

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# Case 3, Conclusions

In case that there are large period time-sensitive traffic streams with their average bandwidth satisfying the admission control criteria, these streams do still cause large bunching delays

 Our simulation results show that the end-to-end delay can be larger than the 2ms objective

- To guarantee the desired end-to-end delay performance with feasible admission control and scheduling and reasonable network efficiency, a method to describe and normalize the time-sensitive traffic should be defined.
  - •This can be done through a protocol adaptation layer (PAL) specification.
  - •The PAL specifies the general properties of time-sensitive traffic carried by ResE. It also specifies how application data are adapted into ResE frames
  - •The PAL for ResE could be similar to IEC61883, and be defined somewhere outside IEEE802.

# Summary

□We showed that to achieve the required delay/jitter/wander performance of transporting audio/video applications on Ethernet, following extensions are needed for existing Ethernet :

- Accurate timing synchronization
- Subscription and admission control
- Protocol adaptation layer definition

### References

- 1. Alexei Beliaev, *Latency Sensitive Application Examples*, Gibson Labs, part of *Residential Ethernet Tutorial*, IEEE 802.3 meeting, March, 2005.
- Geoffrey M. Garner and Felix Feng, Delay Variation Simulation Results for Transport of Time-Sensitive Traffic over Conventional Ethernet, Samsung presentation at July, 2005 IEEE 802.3 ResE SG meeting, San Francisco, CA, July 18, 2005.
- 3. Geoffrey M. Garner, *End-to-End Jitter and Wander Requirements* for ResE Applications, Samsung presentation at May, 2005 IEEE 802.3 ResE SG meeting, Austin, TX, May 16, 2005.