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Delivering Broadcast-Quality audio over unmanaged IP links
The good, the bad, and the perfect

Presenter: Tony Peterle

Email: peterle@worldcastsystems.com

SURESTREAM

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Presentation agenda

- **IP-Audio over unmanaged Networks**
- IP-Audio over the Internet
- SureStream Technology
- SureStream Available on
  - WorldNet Oslo
  - Horizon Next Gen
  - 1U Oslo and AOIP Card

IP-Audio over unmanaged networks

- For many years synchronous technologies have been considered as the backbone technology.
  - T1
  - ISDN
- Migration from synchronized circuits to IP infrastructures is not always embraced by broadcasters, however compelling reasons exist:
  - - constantly increasing costs of synchronous circuits,
  - - significant higher flexibility and scalability on IP infrastructures,
  - - Widespread availability
  - - Network efficiency (multicast, multiple unicast)
  - - Simplicity
- More broadcasters migrating to IP topologies

IP-Audio over unmanaged networks

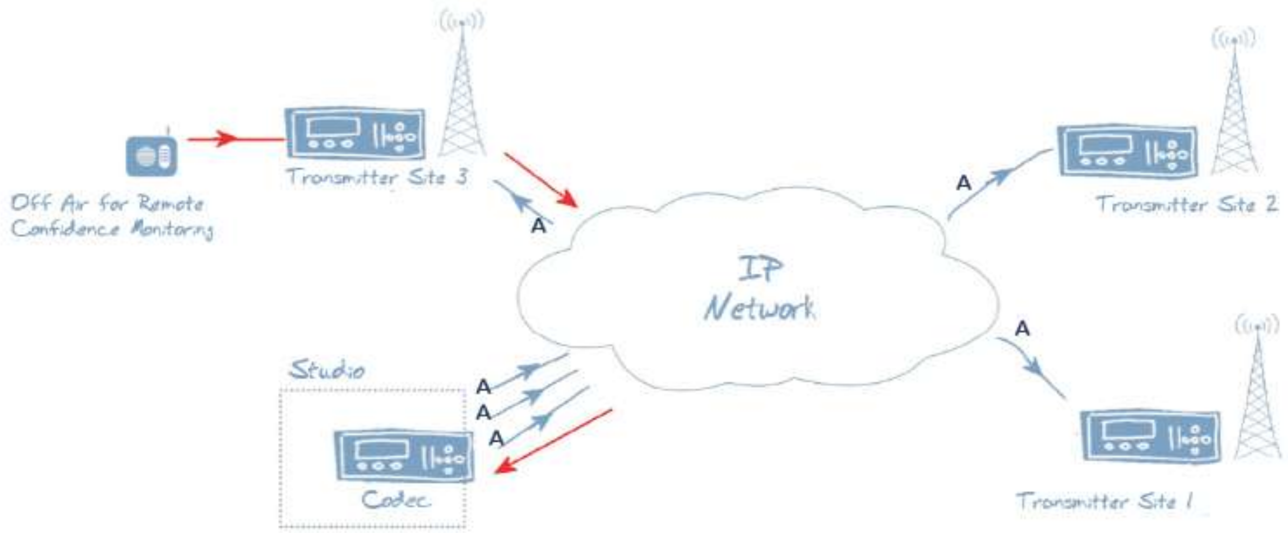
- The migration will happen despite several well established advantages for synchronous links

- The suitability of synchronous links for broadcasters are due to:
  - - reliability
  - - point-to-point bi-directional communication
  - - guaranteed data and error rates
  - - fixed and low latencies

- IP era has firmly arrived and is here to stay….

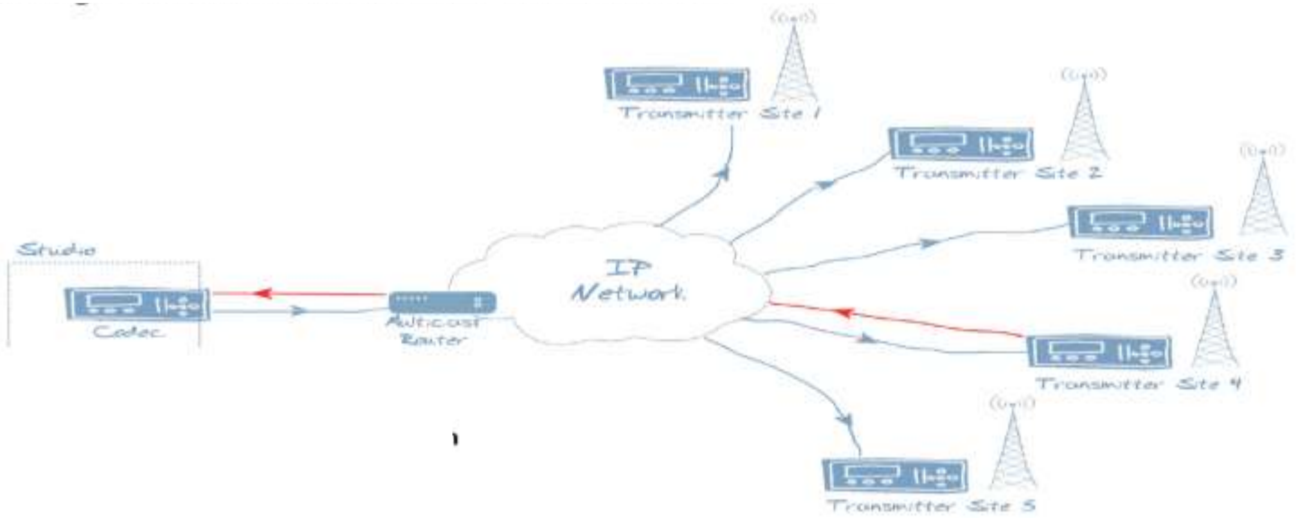
Unicasting or Multicasting ?

- Unicast for simple point to point links.
- Multiple Unicast used to stream from a single site to multiple sites.
- Caution, replicates the bandwidth so bandwidth hungry



Unicasting or Multicasting 2

- Multicast efficient when you need to transport audio from a single point to multiple end points.
- Source codec sends the IP packets to a multicast router using a Multicast Group address as its IP destination address
- Specific network configuration required for multicasting



IP-Audio over managed networks

- Moving to a **managed IP network** allows to put traffic shaping procedures in place. This provides a more stable network:
  - Prioritizing traffic that transport streaming content (audio)
  - Quality of Service (QoS) provides a mechanism to create a hierarchical level for each type of traffic
  - FEC mechanisms achieve redundancy up to a certain degree
  - Dedicated links – uncontended or low contention, no bandwidth sharing
  - MPLS - Multi-protocol Label Switching.
    - Connection-oriented service with ability to support bandwidth reservation and service guarantees.
    - MPLS assigns labels to each packet so the router can switch the packet without reference to the IP address. Supports multiple classed of service allowing risk free sharing of bandwidth.

IP-Audio over unmanaged networks

- **Unmanaged** IP network elements (routers, switches) will apply a simply “best effort” approach to traffic forwarding and will provide no other prioritization (typ.: Internet) – different set off challenges....
- Real Time traffic will be adversely affected with the result of:
  - stream interruption (buffer under run)
  - lost packets (spiky drop out)
  - large swings in jitter performance
  - and finally by an LOC event (Loss of Connection)
- One of the major source of packet loss is created when packets get stuck in a queue that cannot reach the destination in time.
- <u>Transmitting broadcast audio reliably over unmanaged public networks like the Internet needs a very pragmatic solution</u>!

Packet Construction & Size

- Encapsulation into an IP packet adds Ethernet and IP header bytes which contain routing information.
- The size of the headers is constant.
- All the protocol information is contained here, RTP, UDP, SIP, SDP.
- Every packet emanating from a codec to a network must contain header information.
- Packet headers require bandwidth, hence there is a correlation between packet size and bandwidth requirements.

Packet Construction & Size

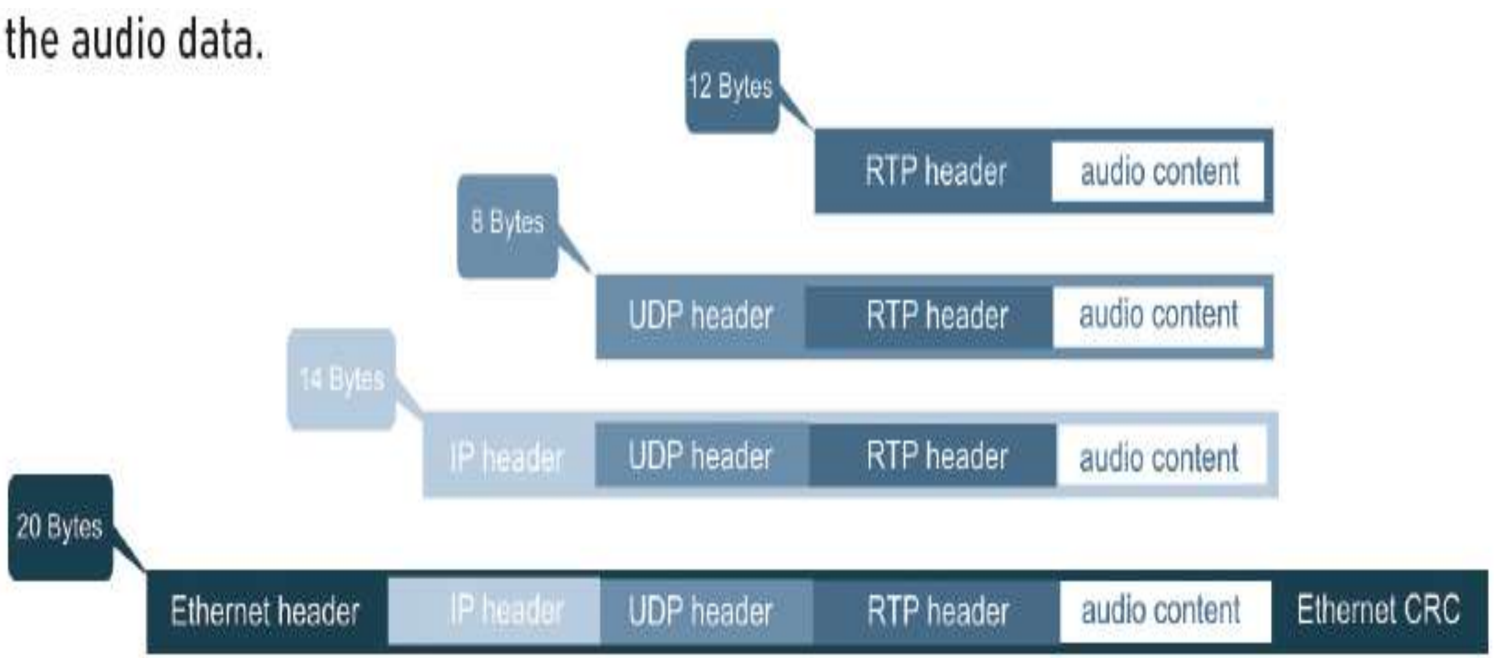


Figure 2: How an IP packet is constructed

Packet Size - tradeoffs

Audio Data Rate	Audio Packet Size (bytes)	IP Packet Size (bytes)	IP Packets/ sec	Packetization Delay (ms)	IP Data Rate
64 kbps	128	194	62.5	16	97 kbps
	256	322	31.25	32	80.5 kbps
	512	578	15.625	64	72.3 kbps
	1280	1346	6.25	160	67.3 kbps
128 kbps	128	194	125	8	194 kbps
	256	322	62.5	16	161 kbps
	512	578	31.25	32	144.5 kbps
	1280	1346	12.5	80	134.6 kbps
256 kbps	128	194	250	4	388 kbps
	256	322	125	8	322 kbps
	512	578	62.5	16	289 kbps
	1280	1346	25	40	269.2 kbps
384 kbps	128	194	375	2.7	582 kbps
	256	322	187.5	5.3	483 kbps
	512	578	93.75	10.7	433.5 kbps
	1280	1346	37.5	26.7	403.8 kbps
576 kbps	128	194	562.5	1.8	873 kbps
	256	322	281.25	3.6	724.5 kbps
	512	578	140.625	7.1	650.3 kbps
	1280	1346	56.25	17.8	605.7 kbps

Figure 3: Table Showing relationship between IP bandwidth requirements, packetization delay and data rates

Network Jitter? Stop drinking so much coffee....

- A characteristic of packet switched networks that every packet can take any route from source to destination.
- Jitter occurs when packets arrive either side of their predicted arrival time.
- Jitter Buffer to store packets so they can be reconstructed and played out coherently.
- Jitter buffer is measured in milliseconds and will affect latency between codec endpoints.

What If I Can’t Get a IP Link to My Transmitter Site ?

- IP Over Microwave RF links has proven to be reliable and robust.
- Licensed and Unlicensed (Typically 5.8GHz).
- Path calculations must be accurate to ensure bandwidth thresholds are acceptable.
- Error rates on the microwave path must also be within a defined threshold.
- Ensure the microwave provider has experience or at least an understanding of the real-time applications.

- **BUT** – what if a microwave IP solution is not possible? (distance, terrain, interference)

The big, bad, scary Internet

- IP-Audio over unmanaged Networks
- **IP-Audio over the Internet**
- SureStream Technology
- SureStream Available on
  - WorldNet Oslo
  - Horizon Next Gen

The Internet is public, and anything public can be messy…

- The Internet does not provide any service quality
- A codec can host protection mechanisms counteracting the network behaviour:
- - FEC forward error correction
  - FEC schemes can protect the content against individual packet losses
  - if configured efficiently, it adds a huge amount of delay
  - increases (multiplies) the data traffic
  - is useless on losses of clusters of packets
- - Dynamic stream adaptation using RTCP
  - - decreases dynamically the audio quality (down scaling)
  - - audible swings of buffer sizes (audio pitch up/down)
  - - precludes a constant link latency
- SureStream adopts a very different approach….

A path through the Internet forest

- IP-Audio over unmanaged Networks
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SureStream - Technology

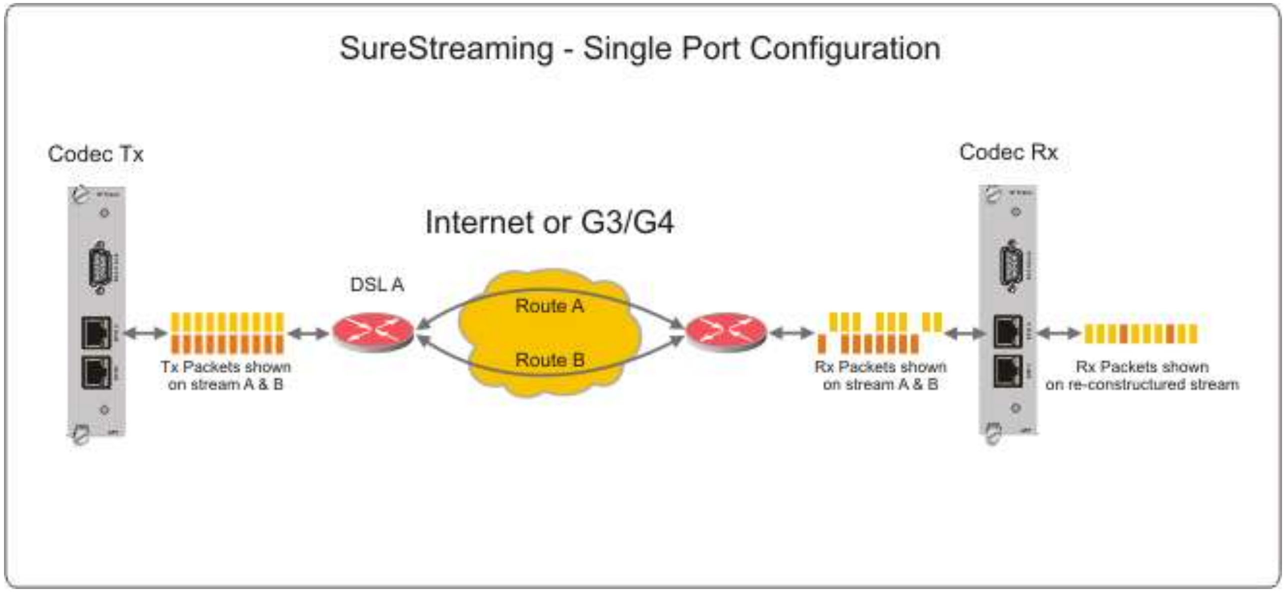
- SureStream Technology is a pragmatic approach: IP-stream-diversity
  - - allows flawless audio transmission over unmanaged networks
  - - affords a constant link delay (fixed latency)
  - - balances large swings of delay jitter (no dynamic buffer size adaptation)
  - - copes with the "best effort" transmission approach of public networks
  - - injects no significant additional latency (unlike FEC)
  - - no restriction on audio formats
  - - flexible by configuration (no restrictions on IP or audio settings)
  - - allows stream protection on multiple levels (as many redundant streams as applicable)

SureStream - technology

- SureStream Technology is based on redundant packet streaming
  - it capitalize the “best effort” transmission approach of the InternetSureStream takes the disadvantage of the un-predictable routing in the Internet and turns it into reliability
- SureStream is flexible by design
  - IP settings are not restricted (IP addressed, UDP ports, packet sizes, delay jitter buffer sizes)
  - Number of redundant streams are limited by codec implementation only (i.e., WN Oslo allows up to 23 redundant streams – not practical)
  - SureStream is already highly efficient with two streams
  - Allows three different sets of redundancy parameters (adaptive to network behaviour)
  - Number of network accesses limited by the codec (one/two or more)

SureStream - single network provider

- SureStream – Redundant Streaming
  - Copes with network packet losses
  - Increases the link reliabilty significantly
  - Developed for unmanaged networks (Internet, 3G/4G)

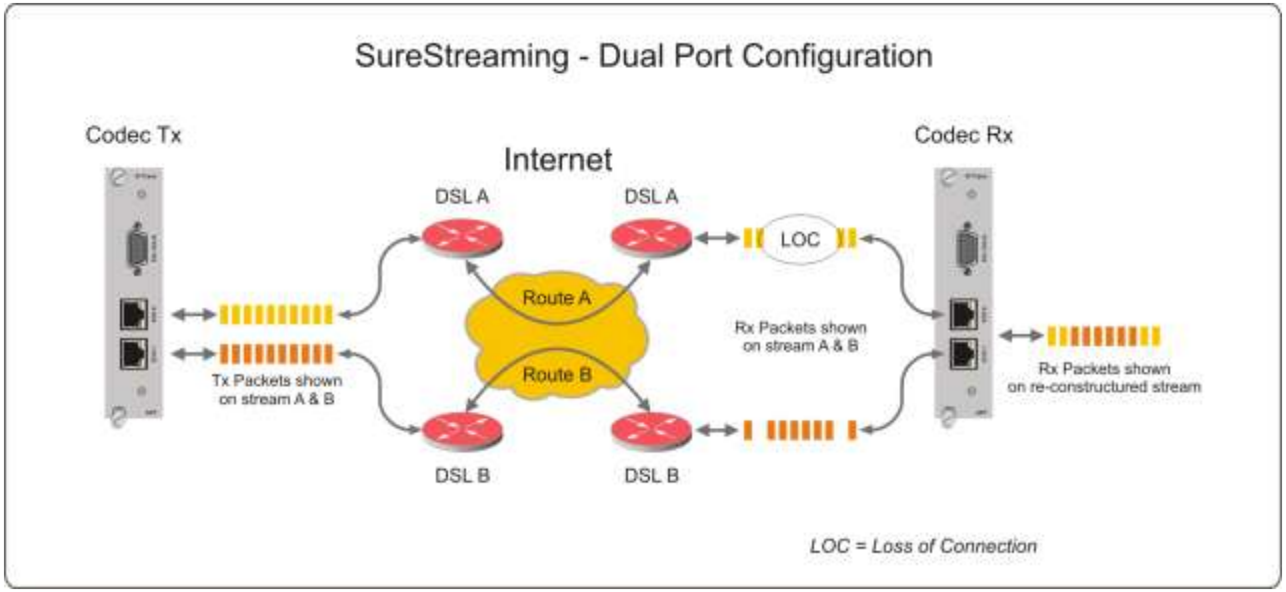


SureStream - technology

- Building blocks:
  - Generating one or more redundant IP streams with identical RTP content on the Encoder
  - Stream shaping by APT's SureStream Engine (SDG Algorithm) on the Encoder
  - Delivering the streams to network, on one or more providers
  - Stream receiver/combiner (ART Algorithm) on the Decoder
  - Delay jitter compensation buffer (ART Algorithm) on the Decoder
  - Packet re-sequencer on the (ART Algorithm) Decoder

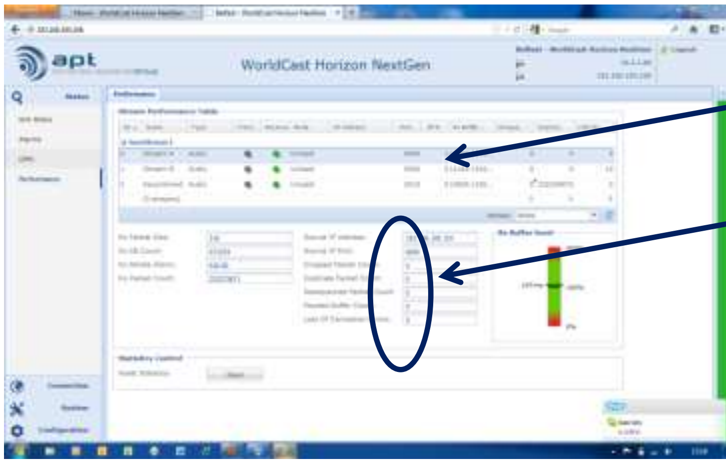
SureStream - dual network providers

- SureStream – Redundant Streaming (dual network access)
  - Copes with network packet losses AND LOC errors
  - Increases the link availability up to 99.99999%
  - Developed for managed (MPLS) and unmanaged networks (Internet, 3/4G)



The SureStream Demo Explained

- One month on the bench with multiple disconnects



Component Stream A
Performance Metrics:
Received Packets 232,273,871
Dropped Packet 0
Duplicate Packet 0
Reseq. Packet 0
Flooded Buffer 0
LOC 5

Component Stream B
Performance Metrics:
Received Packets 232,272,546
Dropped Packet 0
Duplicate Packet 0
Reseq. Packet 0
Flooded Buffer 0
LOC 10

The SureStream Demo Explained

- One month on the bench with multiple disconnects

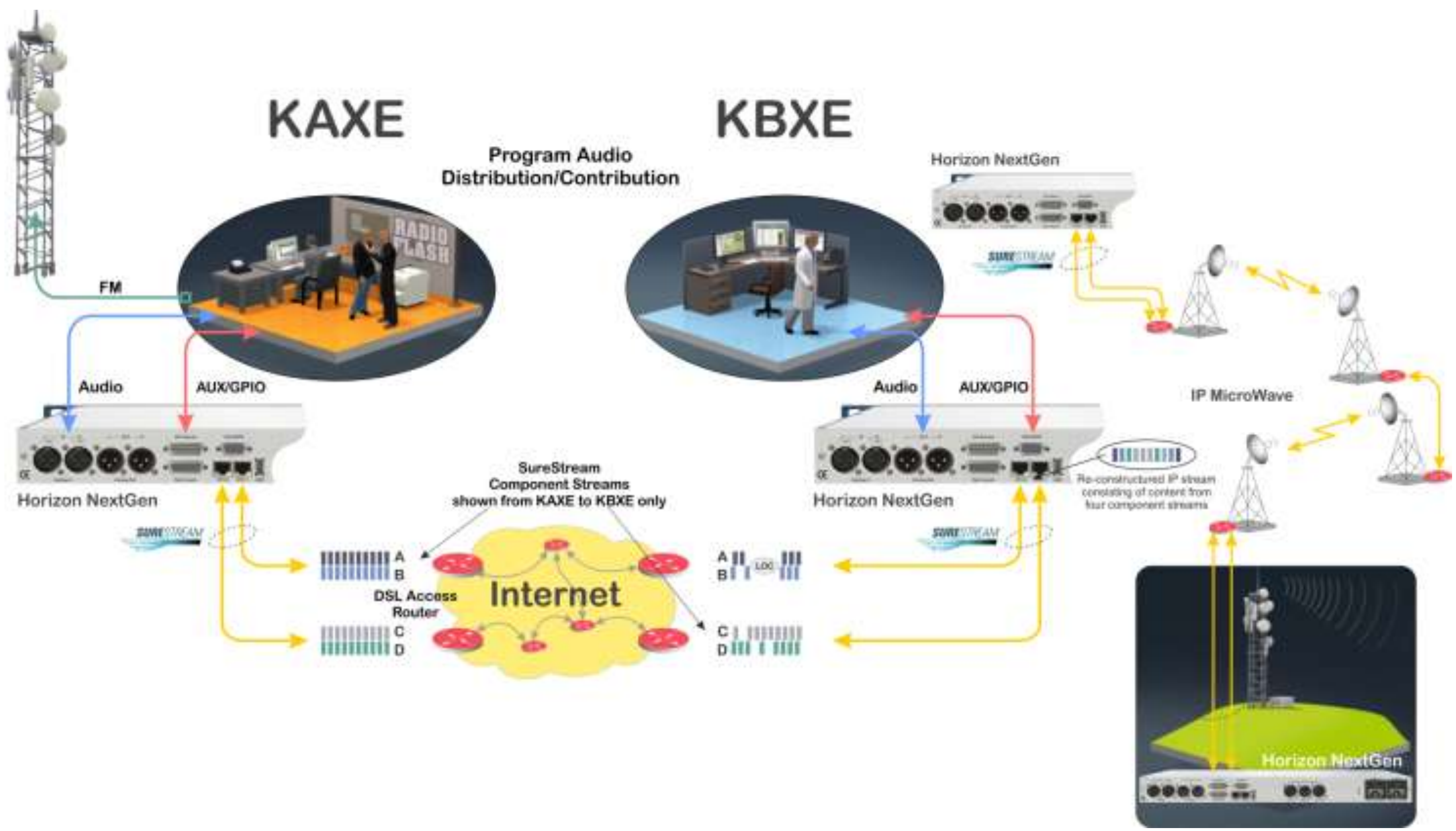
Recombined or SureStream
Performance Metrics:
Received Packets 464,554,906
Dropped Packet 0
Duplicate Packet 232,267,657
Reseq. Packet 0
Flooded Buffer 0
LOC 0

- Multiple disconnects have no affect on recombined or SureStream
- Always on redundant approach proved on lossy networks
- Makes traditional “Main and Backup” redundant options obsolete

SureStream Deployment –KAXE / KBXE Details

- Needed to satisfy two requirements SSL between KAXE / KBXE and STL for KBXE
- Between the two studios installed four consumer grade ADSL, two at each site
- Provider was Paulbunyan.net, links averaged 7Mbits Downlink and 4Mbits Uplink
- Total cost was $240 USD per month, links also used for other IP traffic, not just the SSL,
- Links also used for talkback on program handovers , delay is respectable for public internet at 300ms

SureStream Deployment–KAXE / KBXE Details



SureStream Deployment–KAXE / KBXE Details

- For the second application the STL for KBXE no Telco connectivity at all
- IP connectivity was established with microwave links, :
  - Ubiquity 2.4GHz Nanobridge units, 100 Mbps, 4 Block Link
  - Motorola PTP800 radios, bandwidth limited to 10 Mbps, 17 miles link
- Multiple copies of the content of feeder streams sent on diverse microwave routes, recombined at the far end using SureStream technology

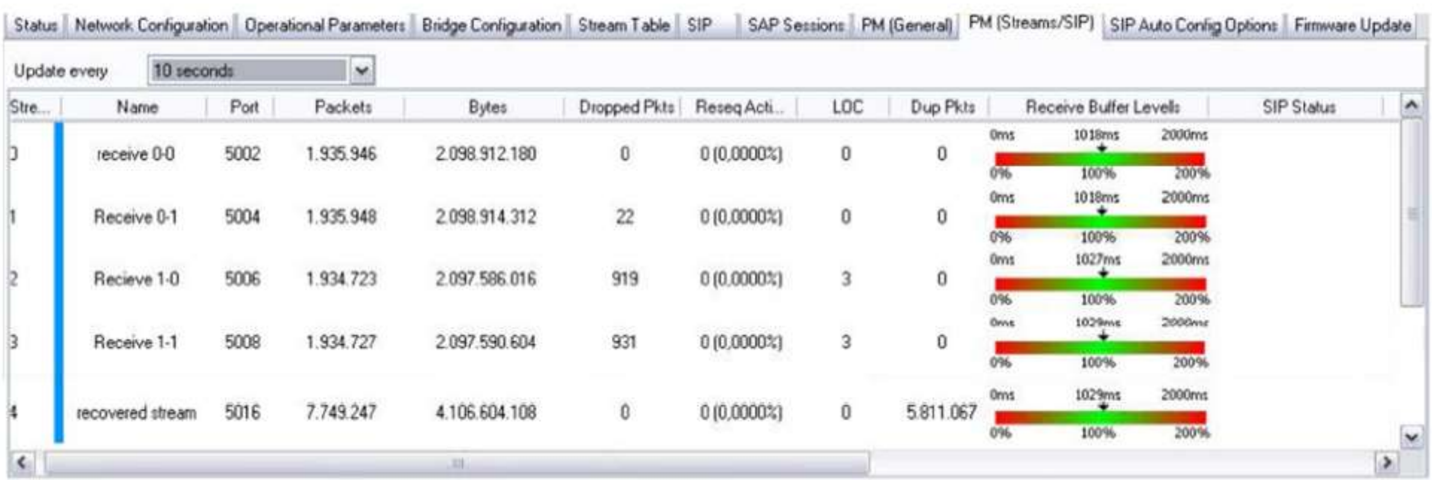
"The SureStream technology has made what was conventionally assumed as impossible possible, that is having a high quality, real-time studio link over the open internet." – Dan Houg, Chief Engineer KAXE / KBXE

Some Other Notable Successes To Date

- WDR (ARD Group, Public Broadcaster in Germany)
  - 5.1 Surround Sound Contribution across city from classical music venue
  - 5.1 Contributions are now economically feasible again
- NDR (ARD Group, Public Broadcaster in Germany)
  - Contribution link from the world athletics championships in Seoul, Korea into main studios in Hamburg, Germany
  - A / B'ed against leased link, audio quality exactly the same, cost -90%
- STL for American MRBI between NYC and NJ (Horizon Next Gen), Cable Modem and ADSL
- Radio Flaix
  - Contribution Link between main studio in Madrid and regional studio in Andorra
  - 1 Fiber optic link and 1 SDSL link terminating each frame, two streams configured per link

Radio Flaix Test – 12 Hour Screen Shot -29/07/11

Date	Algorithm	Stream 1 (Drop / LOC)	Stream 2 (Drop / LOC)	Stream 3 (Drop / LOC)	Stream 4 (Drop / LOC)	**SureStream (Drop / LOC)**
28/07/11	E16 (256kbps)	Test Start	Test Start	Test Start	Test Start	**Test Start**
29/07/11*	E16 (256kbps)	0 / 0	22 / 0	919 / 3	931 / 3	**0 / 0**
02/08/11	E16 (256kbps)	11/ 8	224/ 0	6146 / 8	6283 / 8	**0 / 0**
03/08/11	E16 (256kbps)	11 / 13	261 / 0	7050 / 8	7224 / 8	**0 / 0**


Status | Network Configuration | Operational Parameters | Bridge Configuration | Stream Table | SIP | SAP Sessions | PM (General) | PM (Streams/SIP) | SIP Auto Config Options | Firmware Update

Update every 10 seconds

Stre...	Name	Port	Packets	Bytes	Dropped Pkts	Reseq Acti...	LOC	Dup Pkts	Receive Buffer Levels	SIP Status
0	receive 0-0	5002	1.935.946	2.098.912.180	0	0 (0,0000%)	0	0	0ms 1018ms 2000ms 0% 100% 200%	
1	Receive 0-1	5004	1.935.948	2.098.914.312	22	0 (0,0000%)	0	0	0ms 1018ms 2000ms 0% 100% 200%	
2	Recieve 1-0	5006	1.934.723	2.097.586.016	919	0 (0,0000%)	3	0	0ms 1027ms 2000ms 0% 100% 200%	
3	Receive 1-1	5008	1.934.727	2.097.590.604	931	0 (0,0000%)	3	0	0ms 1029ms 2000ms 0% 100% 200%	
4	recovered stream	5016	7.749.247	4.106.604.108	0	0 (0,0000%)	0	5.811.067	0ms 1029ms 2000ms 0% 100% 200%	

Radio Flaix Test - Continued

- In the 6 days the test was run, there were 14546 dropped packets and 29 LOC, on the 4 streams from Barcelona to Andorra.
- The LOC errors will, as a minimum, drop the entire content in the buffer. In the test, we have roughly 32 packets in the buffer.
- The final result is: 32*29+14546=15474 dropped packet, which gives an average of 3868.5 dropped packets per stream.
- The Oslo unit in Andorra has been, with the SureStream technology, able to recover all the dropped packets.

- Perfect audio over imperfect IP

Belfast to Miami link, stats as of March 6, 2012

- Stream had been up and running since November 2011.
- Basic business DSL connections, 2x on each end.
- Last reset in February, 2012 (ISP change in Belfast).
- Three streams configured, over 500,000 lost packets total.
- Recovered stream – 35 lost packets

Stre...	Name	Port	Packets	Bytes	Dropped Pkts	Reseq Acti...	LOC	Dup Pkts	Receive Buffer Levels	Tx Pkt Size
0	Stream 0-0 Lvl Off	5004	43,192,016	2,050,462,596	36,092	0 (0.0000%)	7	0	0ms 773ms 1500ms / 0% 100% 200%	0
1	Stream 1-1 Lvl Off	5006	40,866,912	3,229,373,220	517,263	1 (0.0000%)	1,930	0	0ms 774ms 1500ms / 0% 100% 200%	0
3	Stream 0-1 Lvl III	5010	43,005,281	1,800,203,964	9,931	2 (0.0000%)	169	0	0ms 769ms 1500ms / 0% 100% 200%	0
4	Recovered Stream	5012	125,071,401	114,352,932	35	427 (0.0003%)	2	81,827,230	0ms 770ms 1500ms / 0% 100% 200%	0

- 35 packets lost/43 million packets in recovered stream = 99.9999%
  Perfect audio over open transatlantic Internet with a 750 ms buffer!

Belfast to Miami link, part deaux

- In March 2012 the Oslos on the link were replaced by 2 Horizon Next Generation codecs.
- Same basic business DSL connections, 2x on each end.
- Statistics last reset in mid September, screenshot captured October 3
- 104,922,829 RX – 78,643,574 Duplicate = 26,279,255 Payload
- 25,165 dropped packets, 138 LoC events
- Recovered stream – **zero losses**

Stream Performance Table

ID	Name	Type	Trans...	Receive	Mode	IP Address	Port	ETH	Rx Buffer ...	Droppe...	Duplica...	LOC Er...
SureStream 1												
0	Port 0-0 Lvl...	Audio			Unicast		5010	0	25284 (105....	3255	0	16
1	Port 1-0 Lvl...	Audio			Unicast		5020	0	29136 (121....	21629	0	118
2	Port 0-1 Lvl...	Audio			Unicast		5030	1	25284 (105....	281	0	4
3	Port 1-1 Lvl...	Audio			Unicast		5040	1	25680 (107...	0	78643574	0
	(4 streams)									0	0	0

Belfast to Miami link, part deaux

Stream Performance Table

ID	Name	Type	Trans...	Receive	Mode	IP Address	Port	ETH	Rx Buffer ...	Droppe...	Duplica...	LOC Er...
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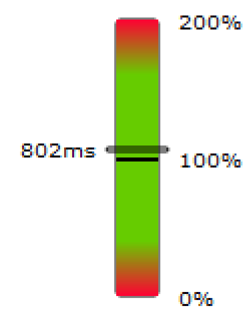
3	Port 1-1 Lvl...	Audio			Unicast		5040	1	25680 (107...	0	78643574	0
	(4 streams)									0	0	0

Refresh : 10s

Rx Packet Size:	1284
Rx kB Count:	3511694
Rx Bitrate (kb/s):	1048.55
Rx Packet Count:	104922829

Source IP Address:	87.192.228.245
Source IP Port:	6000
Dropped Packet Count:	0
Duplicate Packet Count:	78643574
Resequenced Packet Count:	38
Flooded Buffer Count:	0
Loss Of Connection Errors:	0

Rx Buffer level



Award Winning Technology – The accolades so far…

Presentation [agenda – 1/4]

- IP-Audio over unmanaged Networks
- IP-Audio over the Internet
- **SureStream Technology**
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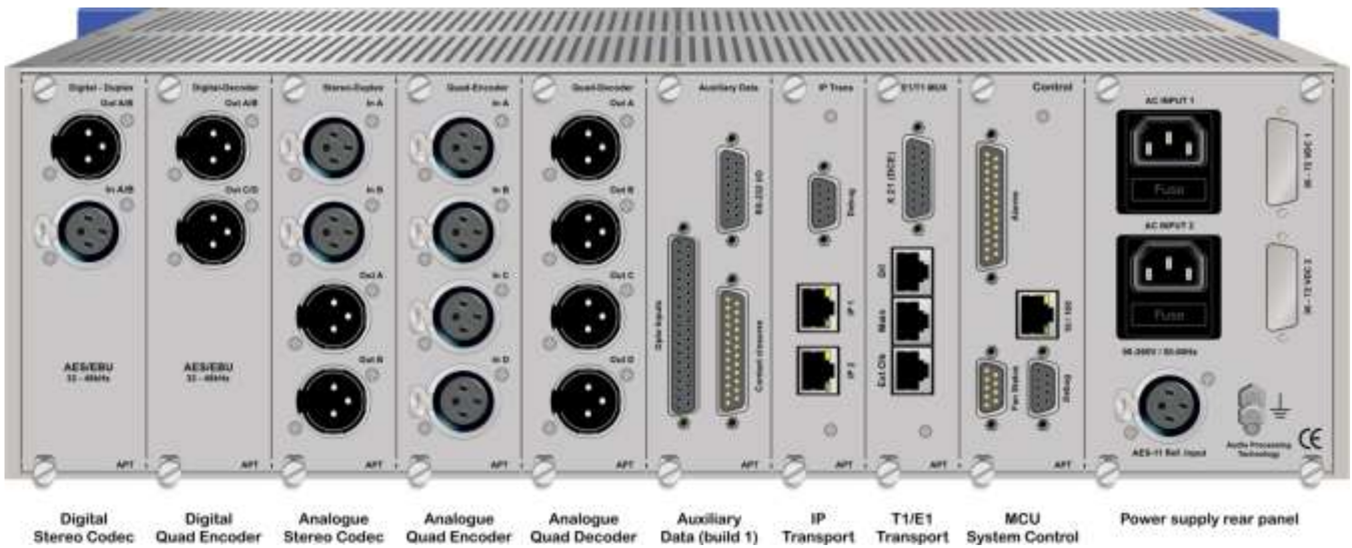
WorldCast Horizon Next Gen

WorldCast Horizon Next Gen

- Hardware
  - DSP based
  - analogue and digital inputs and outputs on XLR
  - AES reference input
  - 4 Opto coupled Inputs and 8 Relay Outputs
  - Aux data port – RS232
  - front panel, power connection and alarm LEDs
- Options
  - redundant PSU
  - second IP port (control /streaming)
  - front panel meters, headphone socket and monitor switch
  - SureStream Software

WorldNet Oslo

WorldNet Oslo



- WorldNet Oslo Rear Panel view:
  - Variety of different audio Encoder/Decoder/Codecs
  - AUX Data payload card
  - IP and TDM (E1/T1) transport card

Oslo 1U and AOIP Card

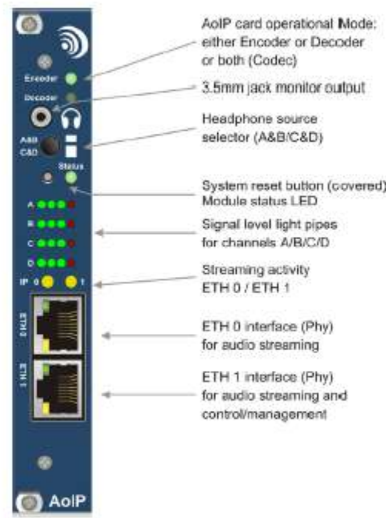
Technical Specification

Physical	Size	3U x 19" Rackmount
	Dimensions	133mm x 482mm x 430mm 5.25" x 19" x 17"
	Weight	9 kg / 19.8 lbs
	AC Power Supply	90 - 260 VAC, 47 - 60 Hz
	DC Power Supply	36 to 72 V DC
	Power Consumption	<200 W
	Environmental	+ 5° C to + 45° C
IP	IP	Complete IP-engine including buffers
	Dual 10/100/1000 BaseT interface	Audio & control, second port audio only
	Clock	VCXO per mono channel
	Modes	Multicast (IGMPv3), unicast, multiple unicast
	VoIP	SIP, STUN, DHCP, SureStream (APT), SmartIP (APT)
	Security	QoS: DiffServ, FEC (SMPTE 2022)
	VLAN Tagging	Per individual stream (IEEE 801q & 1p)
Audio	Audio Input / Output	Analogue and AES/EBU
	Sampling Frequencies	32 kHz to 48 kHz
	Audio Bandwidth	10 Hz - 22.5 kHz
	Analogue Mode	Balanced
	I/P Impedance	>24k/600 Ohms, Symmetrical
	O/P Impedance	<100/600 Ohms, Symmetrical
	Digital Mode	Balanced

	Impedance	110 Ohms / 75 Ohms
	Digital Ref In	Balanced
	Dynamic Range	16 bit > 85 dB, 24 bit > 110 dB
Coding	Transparent AES	Including C, U and V bits
	Linear PCM	16, 20 and 24 bit
	Enhanced apt-X	16 and 24 bit, 64 kb/s to 576 kb/s
	MPEG I/II Layer 2	
	MPEG Layer 3	VBR / CBR
	MPEG 4 AAC	LC, LD and HEv2
	G codes	N/ACIP Tech Doc. 3326 v3
Data	Aux Data	Up to 2 channels per card RS232 / RS422
	Aux Data Mode	Embedded or non-embedded
	Data Rates	1200, 2400,4800, 9600, 19200 Baud
	GPIO per AoIP card	4x switch inputs and 4x relay contact closures
Control	Management	Network Management System, Web GUI, SNMP
	MCU	Central control of PreSets, Backup, Logs, etc
	Control In MCU	10/100 BaseT Ethernet (RJ45)
	Alarms	15 pin D type, 7 Relays, 3 Contacts Per Relay
	SNMP	Version 1, 2c and 3

Oslo 1U and AOIP Card

- Fully independent N/ACIP compliant AoIP Module
- DSP based, high density audio solution
- Universal hardware for analogue/digital mode selection
- Audio card provides four signal paths:
  stereo duplex, quad encoder, quad decoder
- Independent encoder/decoder configuration and automatic detection of the RX audio mode
- Main board configuration follows I/O board



Summary

- SureStream allows for huge savings on OPEX
- SureStream means no compromise on audio quality
- SureStream provides unsurpassed redundancy over open internet
- Scalable building block of products, stereo 1U to multichannel 3U units
- Low delay over IP combining Linear or Enhanced apt-X with SmartIP codec engine (all broadcast applications covered)
- Enhanced apt-X and Linear transparent delivery, protection against audio quality issues generated further down the broadcast chain (concatenation)
- Range of redundant options, PSU, Transport (IP to IP, IP to ISDN etc), Audio (SD Card Backup and N+1) – ethos, "stay on air no matter what"
- Among the best MTBF figures in the industry, DSP reliability for 24/7/365 operation

One More Thing........

- In order to properly acknowledge my gratitude for the opportunity to speak before the Wisconsin Broadcasters, I felt it best to consult a local expert
- Fortunately, at the Orlando Ennes conference on Friday, just such an expert was available…….

- And he attempted to teach me the “Bucky”

Thanks for your attention!
Questions?

Presenter: Tony Peterle

Email: peterle@worldcastsystems.com