BT-HMP

Multimedia processing for next generation

BRIDGETEC

As telephone network change from TDM-based to IP-based, new opportunities arise for value-added service providers and developers. Creating new services to the market required significant investment of time and equipment for the existing TDM-based network. Exclusive DSP/E1 board was needed, and stability and load tests were time consuming. Also, more and more space and power are needed when service is to be extended. With just one server and corresponding technology, voice, video, and fax services can be provided in IP network. Today host server density equals or surpasses the DSP board because of breakthrough in CPU performance.

If you have technology and idea, you can provide services with almost no initial investment. Developers must decide which company's HMP (Host Media Processing) software to use. Let's consider the points in choosing the HMP software.

- Should support audio, video, and fax in this multi-media era.
- Should Provide higher performance on same specification.
- Should use less resource to operate in hardware with low capacity.
- Should have security features.
- Should provide high quality audio/video

1. Multi-Media Support

Following codecs should be supported for audio services.

- g.711 alaw/ulaw
- g.723.1
- g.729

Though compression codecs such as g.723.1 and g.729 are becoming less important as IP network gets wider bandwidth, these codes are still necessary because if you use dedicated line, the cost per byte is very high.

Voice prompt playing and recording capabilities are required for each codec. If audio saved in files and packets transmitted by IP network have different codec, they need to be transcoded. Creating audio files using g.723.1 or g.729 and native playing/recording reduces load significantly because transcoding g.723.1 and g.729 consumes much CPU time.

The ability to switch two calls connected to the server is required. Typically, decoding audio streams to PCM takes place before switching and encoding by IP packet codec. If two calls have same codec, using native switch reduces the load.

DTMF recognition capabilities of in-band method and rfc2833 method are required. Especially when two methods are used together, the DTMF should not be recognized twice.

Following codecs need to be supported for video services.

- H.263
- H.264
- Mpeg4

A variety of play features are required for video services. Various features such as text/Image overlay and scroll need to be supported. Scrolling horizontally and vertically need to be supported, and the ability to specify font and area on the screen should be supported when overlay is used.

Bitrate control is needed to provide services in limited bandwidth such as 3G networks. Particularly bitrate tends to sharply increase when transmitting intra frame; how to control this is the key.

T.38 is used for fax services in IP network. One must consider whether ECM mode is supported, what the maximum transmission rate (v.17 14,400 bps, v.34 33,600 bps) is, what compression methods (MH, MR, MMR) are supported, and what resolutions (standard, high, super high) are supported.

2. Performance

Typically g.723.1/g.729 requires as much as 2-3 times the load than g.711 for voice services. However, using native method eliminates the difference in loads. Decreasing frame size (which means more packets are being transmitted) consumes more loads. So the frame size becomes an important factor in g.711. Using Secure RTP consumes more loads because the process of encrypting and decrypting packets is added. In the case of g.711, the load increases by approximately 30%.

Image size is an important factor for performance in video services. The length and width of CIF are twice as long as QCIF, and CIF requires load twice more than that for QCIF. H.264 doubles the quality of h.263 in same bandwidth but consumes CPU load twice as much.

Recent servers are equipped with at least a dual core CPU. It is also important if HMP is designed to perform optimally in multi-core environment.

3. Using Less Resource

It is better if less resource is used because audio, video and fax services and user applications run in one server.

Resource means the amount of memory used. Recent servers are typically equipped with several giga bytes of memory, and much of the memory should be taken up by the applications.

4. Security

SRTP should be supported to prevent wiretapping, monitoring, or tampering of audio packets.

The capability to filter randomly received invalid RTP packets when SRTP is not used, should be provided.

5. QoS

HMP-method services are particularly vulnerable to QoS because HMP software operates with other applications in general-purpose OS. HMP needs to operate in real time to transmit and receive packets reliably even in overloaded condition.

Latency and delay tends to increase in HMP method services. End-to-end delay should be within 150ms for the optimal audio service especially for conference services where delay increases to the maximum.

How fast lost packets are recovered is the key in video services. Sending intra-frame recovers the screen, but there are standards for when and how to send the intra-frames. Not all devices support all standards, so mutually compatible method should be found.

6. Conclusions

Bridgetec's BT-HMP is designed by such considerations. Also, BT-HMP's intuitive and easy API allows developers to create media services more quickly. Developers will be able to control applications more precisely and can integrate with 3rd party libraries unrestricted, using BT-HMP's flexible and unlimited API functions. BT-HMP's features are described below.

❑ Feature (rtp device)

Function	Description
RTP/RTCP	Support rfc3550
SRTP/SRTCP	Support rfc3711 Support AES, ARIA, SEED, SHA-1
Codec	alaw, ulaw, g.723.1, g.729 h.263, h.264, mpeg4
Dynamic jitter buffer	Start at minimum buffer length and increase when needed to reduce delay
Silence compression	The amount of data sent is reduced or data is not sent if not talking
Packet loss concealment	Improve sound quality by playing intermediate value when packets are lost
Comfort noise generation	When packets are not received, or silence compressed packet is received, making comfort noise makes you feel as if you are on the phone.
File codec for play and record	alaw, ulaw, 16 bit PCM, g.726, vox h.263, h.264, mpeg4
Video image size	QCIF, CIF
Video frame rate	5~30
Video rendering	text/graphic overlay horizontal/vertical scroll paging word wrapping

	DTMF echo
Play	File or memory play Native or transcode play Transmit mix of play data and switch data
Barge-in	Stop playing when DTMF or voice is recognized
Play volume/speed	Adjustable
Record	File or memory record Native or transcode record Record mix of incoming and transmitting data
DTMF recognition/formation	In-band and rfc2833 recognition/generation
Fax tone recognition/formation	CNG recognition/generation
Voice activity detection	Detect starting and ending points
DSCP	Provide marking ability
QoS alarm	Alert in the event of lost packet, excessive jitter

❑ Feature (conference device)

Function	Description
Maximum capacity	30 participants per room
Monitor, coach, pupil	Freely configure in/out topology between participants
Bridge	As low as 40ms audio delay is possible in bridged room
File codec for play, record	alaw, ulaw, 16 bit PCM, g.726, vox
Play	Play to all or specific participants File or memory play
Record	Record all or specific participants File or memory record
Active talker	Ability to recognize who is talking at the conference
Automatic gain control	Adjust the volume of conference participants
DTMF clamping	Select on/off

❑ Feature (fax device)

Function	Description
Maximum speed	v.17 14,400 bps
encoding	MH, MR
ECM	Support
Page retransmission	Support
Bad scan line recognition and replace	Support
Width resolution	200, 300, 400 pel/inch
Length resolution	3.85, 7.7. 15.4 line/mm

❑ Feature (switch)

Function	Description
switch	One-way switch between RTP and conference devices
Transcode switch	Default switch mode
Native switch	Native switch possible between RTP devices
DTMF switch	Properly converts between in-band and rfc2833

❑ Performance

Test HW: quad core Intel(R) Xeon(R) L5408 2.13G * 2

Codec	Frame size	Number of channels	CPU
g.711	10 ms	960	21%
g.729	10 ms	720	57%
g.723.1	30 ms	720	58%

Test HW: quad core Intel(R) Xeon(R) X5450 3G * 2

Codec	Image size	Frame rate	Number of channels	CPU
h.263	QCIF	15	400	50 %
	CIF	30	60	50 %
Mpeg4	QCIF	15	400	50 %
	CIF	30	70	50 %
h.264	QCIF	15	120	50 %
	CIF	30	20	50 %

❑ Security

Function		Description
SRTP	Encryption Algorithms	AES-CM, AES-F8, SEED, ARIA, NULL
	Authentication Algorithms	HMAC-SHA1, NULL
Packet filter		Source address is checked, and RTP packets not received from the relative terminal are filtered

❑ QoS

Function	Description
Dynamic jitter buffer	Start at minimum buffer length and increase when needed to reduce delay
Silence compression	The amount of data sent is reduced or data is not sent if not talking
Packet loss concealment	Improve sound quality by playing intermediate value when packets are lost
Comfort noise generation	When packets are not received, or silence compressed packet is received, making comfort noise makes you feel as if you are on the phone.

video	FIR(fill intra-frame request) RTCP feedback Bitrate adaptation

❑ Installation Requirements

256MByte of Memory for voice service

1GByte of Memory for video service

128MByte of Memory for fax service

500MByte of hard disk space