Abstract:

DSP applications in Voice over Internet protocol

(1) Group Member

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(2) Background and context

The IP phone was born after commercialization of internet. Thus, people want to use this to replace the traditional interactive communications, which were made by telephone at PSTN line. Data exchange was expensive (for a long distance) and no one had been thinking video interactions (there was only television that is not interactive, as known). In June 1996, Study Group 16 of ITU decided on H.323 v.1, referred to as a standard for real-time videoconferencing over no guaranteed QoS LANs. It is the first VoIP standard to assistant the telecommunication industry move to the IP voice area.

In February, 1995, an Israel company, VocalTec, developed a software (internet phone), which could be used to make a long distance call from internet. It opens the history about VoIP technology. Some biggest vendor, Frost & Sullivan, IDC, join the VoIP market since 2001. Most of them use H.323 agreement, a popular agreement in current market survey. A report, conducted by the Economist Intelligence Unit on behalf of vendor AT&T, said 43 percent of respondents were currently using, testing or planning to implement VoIP within the next two years. A further 18 percent said they planned to implement VoIP in the longer term.

(3) VoIP standard and description

There are four standards of VoIP familiar:

- ❖ ITU H.323: A fully standardized voice over IP specification prevailing currently;
- ❖ ITU H.248: Media Gateway Control Protocol, but it doesn't include a voice specification, but it is an essential element in the IP call pathway if more than one vendor's equipment is in use;
- ❖ Telcordia GR-303: it enables the link of the PSTN for voice-over-cable;
- ❖ SIP: Service Initiation Protocol, not just defined for voice service, but used in combination with large number of popular voice. Now it is used to establish calls between corporate IP services.

In this project, only one standard, H.323, is focused on. There are four components specified in this standard, including terminal, gateway, gatekeeper and multiple control units. In VoIP system, a common voice coding standard, G.711, is necessary to be mentioned. G.711 is a 64-kbps PCM voice coding technique, and is the recommendation specified by ITU-T.

(4) Aims

- ❖ To understand the knowledge of Analog-digital and Digital-Analog conversion;
- ❖ To introduce the VoIP technology and system;
- ❖ To investigate a codec standard-G.711 in VoIP;

(i.e. a 64-kbps PCM voice codec technique for VoIP system)

- ❖ To investigate some techniques of VoIP digital signal process;

(Including QoS (Quality of Service) techniques; a technique to remove the background noise (option); a technique to "introduce" some background signal into the communication (option) ;)

- ❖ To explore the aggregation of VoIP streams in CDMA network (optional);

(5) Responsibility of group member

We have no definite division of responsibility, therefore we plan to share all the materials, compose the report and do the presentation together.