Improved Communication through VoIP / Chat for a distributed pair programming tool (Saros)

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Overview

- Motivation
- Requirements
- Audio Codec comparison (Speex & iLBC)
- Use Cases
- My Implementation
- Timeline

Motivation

- Easy communication in Saros without external applications
- Everyone who is in a shared project in Saros should have the possibility to communicate with each other member of the shared project (Chat & VoIP)
- Some logging options (e.g. the time someone is speaking / chatting, percentage of Saros sessions in which VoIP or Chat was used)
- Possibility to record the VoIP / chat conversation by the members of the session

Requirements (VoIP)

- VoIP Communication with more than 2 people
- One VoIP session in one shared project
- Everyone in a VoIP Session can invite everyone who is in the shared project
- The user who wants to use VoIP can adjust the codec settings (lower bit rate with IBB transmission)
- IBB, Jingle detection to automatically adjust the codec settings

Requirements (Chat)

- Chat with more than 2 people (Multi User Chat)
- One MultiUserChat (MUC) in one shared project in a separate Chat View
- Everyone in a shared project can join the MUC
- Problem of the actual chat implementation is that everyone who is using Saros is chatting with everyone else in one chatroom

Audio codec comparison

- Speex
- Supports bit rates from 2 kbit/s up to 44 kbit/s
- Supports variable bit rate encoding (VBR)
- Voice activity detection (VAD) (in combination with VBR encoding)
- Supports discontinuous transmission (in addition to VBR and VAD) → the transmission will be stopped when the background noise is stationary
- For example Speex is used in Asterisk, Mumble or Teamspeak

Audio codec comparison

- iLBC (internet low bitrate codec)
- Fixed bitrate (15.2 kbit/s – 20ms frames or 13.33 kbit/s – 30ms frames)
- No VBR support
- High robustness to packet loss

- For example iLBC is used in Skype, Google Talk, Yahoo Messenger or Gizmo5 (SIPphone)

Inviting s.o. to a VoIP Session:

Scope: A new invited user has joined a VoIP Session

Actors: Initiator, another User

Precondition:

- Both clients are in one shared project
- Both have correctly installed and configured record and playback devices (Headset)
- Record and playback device need to be set as default device in system sound options

Success guarantee:

- The user has been invited to the VoIP Session

Use Cases

Success scenario:

1. Initiator changes to the „Shared Project Session“ view
2. Initiator selects the user he wants to invite to a VoIP session. Afterwards he clicks on the telephone button in the „Shared Project Session“ view toolbar.
3. The selected user gets a yes-no popup.
4. User accepts the invitation
5. A new VoIP session will be started or the new user will be added to an existing VoIP session.

If the user doesn‵t accept the invitation the initiator will be informed that his invitation has been rejected.

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Leave a VoIP Session:

Scope: One Client leaves the VoIP Session

Actors: Client

Precondition:
The client is in a VoIP Session

Success scenario:
Client changes to the „Shared Project Session" view
Client clicks on the „telephone_stop" button
Client leaves the VoIP Session
(Other members of the VoIP Session will be informed)

My Implementation

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- VoIP in Saros uses jSpeex (Java Speex API) as Encoder / Decoder (at the moment)

  Problem: The audio playback is stuttering

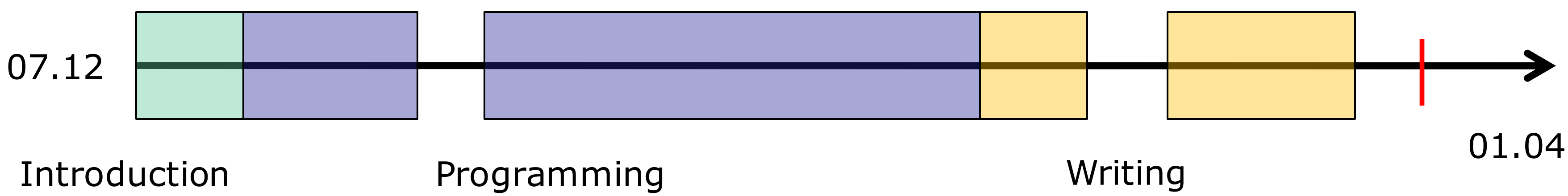
- VoIP uses the StreamService (by Stephan Lau)
- The VoIP function doesn‘t use the XMPP VoIP Interface because of some problems with identification of capture and playback devices. (tested with the Spark IM)
- Existing chat function is using the XMPP Extension for Multi User Chat (XEP-0045)

Timeline

- Week 0 (07.12 – 11.12):
  Introduction to Saros
  Introduction to XMPP
  Introduction to Speex Java API (jspeex)
  Adding already implemented Chat to Saros and testing
- Week 1 (14.12 – 18.12):
  First prototype for audio encoding (record sth. and play it) with JSpeex
- Week 2 & 3 (20.12 – 08.01):
  Simple integration in Saros (with some problems, e.g. stuttering audio playback)
  Registration of the Bachelor thesis (official deadline: 01.04)
- Week 4 (11.01 – 15.01):
  Fixing the stuttering audio playback
  Preparation for the introduction presentation
- Week 5 & 6 (18.01 – 29.01):
  Testing and adding other codecs
  VoIP Session with more than 2 persons
  User Interface

Timeline

- Week 7 & 8 (01.02 – 13.02)
  Adding logging functionality
  Revision of the Chat functionality
  Testing
  Error Handling
  Preparing for Release
- Week 9 (15.02 – 19.02)
  Probably Release week
- Week 10 & 11 (22.02 – 05.03)
  Maybe fixing some bugs
  Starting written thesis
- Week 12 (15.03 – 19.03)
  Finishing written thesis



Sources

- http://fmj-sf.net/
- http://www.ilbcfreeware.org/
- http://speex.org/
- http://jspeex.sourceforge.net/
- http://www.voip-info.org/

Vielen Dank!