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# Innovative Computer-Assisted Meeting Monitoring Module (ICA-3M) with New Sampling Approach

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## Abstract

Technology nowadays evolution brings to the advent of new services, including Voice over IP (Internet Protocol) or VoIP. This new technology serves as the basis for the operation of videoconference-based meeting platforms with the integration of remote participants. In this paper, we propose to contribute by the means of algorithms to improve the functionalities of these platforms in the aspect of a meeting that can involve both face-to-face and remote users. The proposed principle is that of sampling the course of the meeting in order to detect the important data exchanged and processed, to store it in a lighter text format and to automatically generate a meeting report. The sampling grid for the transition from continuous meeting mode to structured mode with useful text recorded is proposed. We define the ICA-3M conventional parameters used by the AT-SAM/DS (Algorithm for Transition from Structured Audio Mode to Data Structures) algorithm for switching to discrete format from the continuous format of the normal meeting flow.

**Keyword:** VoIP, videoconference meeting platforms, method for meeting sampling.

## Introduction

Telecommunication offers many services for the transmission, remote transmission and reception of information of all kinds by wire, radio, optical or electromagnetic systems After the advent of broadband, Technologies evolution brings to the advent of new services, including voice over IP or VoIP. It is a voice communication technology that ensures the convergence of the voice, data and video triplet. However, the interconnection of users in a system requires standards. This is why standard protocols such as H.323 and SIP (Session Initiation Protocol) have appeared. This new technology serves as the basis for the operation of videoconference-based meeting platforms with the integration of remote participants. In this paper, we propose to contribute by the means of algorithms to improve the functionalities of these platforms in the aspect of a meeting that can involve both face-to-face and remote users. The proposed principle is that of sampling the course of the meeting in order to detect the important data exchanged, to store it in a lighter text format and to automatically generate a meeting report.

This paper is organized into several sections. The first recalls the basic principles of a meeting. The second part is devoted to the description of VoIP with the limitations of current platforms. Subsequently, our contribution will focus on the basic principle of ICA-3M (Innovative Computer-Aided Meeting Monitoring Module) with format conversion techniques as well as algorithmic modeling of the course of a meeting. The proposal of the AT-SAM/DS algorithm (Algorithm for Transition from Structured Audio Mode to Data Structures) will precede the highlighting of the Contribution of ASR (Automatic Speaker Recognition) and NAL (Natural Language Processing) to the automation of a meeting before the description of the architecture of the new system.

**Basic principles of a meeting**

Several parameters contribute to defining the basic principles on which meetings are based, even if they can be of several natures. We choose here the model of a faculty council in an educational institution. This type of meeting, like most of those held in societies, concerns an audience known in advance with a fixed number of regular participants. An attendance list is drawn up at the beginning of each meeting. Similarly, the participants speak the same language and the agenda is not excessively long. It is chaired by a presiding officer who distributes the Token for Speaking Autorisation (TSA) in turn, addressing the items on the agenda one by one until they are exhausted. Proposals for solutions to the problem related to the agenda item under discussion and one or more proposals are retained before moving on to the next item. At the end, a summary is made and a report is produced validated by the participants. A discretization approach allows to model the course of these meetings by highlighting its properties and its following articulations:

1. Protractor
a. Establish an attendance list
b. Note Start date and time
2. Chair:
a. Announce Agenda
b. Move forward point by point (item) or by problem to be solved
c. Open the Speaking Lists (SL) by item to be dealt with
d. Distribute the Token for Speaking Autorisation: Alternating Mode

e. Collect the contributions of each actor: several possible contributions or proposals on the same point by the actors present can occur
f. Summarize the contributions by problem to be solved and indicate those selected
g. Agenda exhausted: close the meeting
3. Protractor:
a. Record the end time and calculate the duration
b. Prepare a report

**The VoIP and the Limitations of Current Platforms**

The VoIP technique [1] [2] uses RTP (Real Time Tronsport Protocol) [3] [4] to transmit voice or multimedia communications (video for example) via the Internet. Its role is to organize packets at the entrance and exit of the network for real-time transport and is a protocol suitable for applications with real-time properties. On the other hand, it does not provide a reservation of resources on the network like RSVP (Resource reSerVation Protocol) [5] [6] type action. Although autonomous, RTP can be supplemented by RTCP (Real Time Control Protocol) [7] that provides feedback on the transmission. It adapts a type of coding, modifies the data rate, sends control messages to all session participants and builds a QoS (Quality of Service) report.

The H.323-based VoIP solution : H.323 [8] is a communication protocol encompassing a set of standards used by the VoIP service for sending audio and video data over the Internet. H.323 standardizes procedures for establishing and managing calls and establishes a list of mandatory or recommended audio and video codecs for both parties to negotiate with each other and exchange calls. This protocol is used for real-time interactivity, including video conferencing (signaling, check-in, admission control, transport and encoding). The H.323 protocol stack is independent of the networks and transport protocols used and operates on an end-to-end strategy that gives it transparency to network changes.

The SIP-based VoIP solution : SIP [9] [10] is a signaling protocol belonging to the application layer of the OSI model. Its role is to open, modify and release sessions. To open a session, a user issues an invitation carrying a session descriptor allowing users to agree on the compatibility of their media. SIP therefore allows to link mobile stations by transmitting requests to the current position of the called station. It is based on the HTTP (Hyper Text Transport Protocol) and can use UDP (User Datagram Protocol) or TCP (Transmission Control Protocol). The elements of SIP's protocol architecture are: UDP, TCP, IP, RTP and RTCP complemented by the G711, G722, G728, G729 standards, which are ITU audio compression standards that define the PCM-U and PCM-A encodings based on the quantization law.

Today's face to face and remote meeting platforms use an approach that focuses on audio and video recordings as continuous data streams. This data format storage needs a significant space with a cumbersome time in their remote access. In addition, their repatriation generates bandwidth consumption. It is therefore possible to offer a lean form of storage and availability of this data based solely on the text automatically generated during the meeting. In addition of lightening, this solution also allows to make a significant contribution to the identification of the content discussed during the meeting, with the automatic generation of reports at the end. It is to achieve this objective that we propose to develop the ICA-3M system.

**Algorithmic modelling of the flow of a meeting**

We consider the schema and the nature of the dialogue that is established between the participants of the meeting with a modeling in algorithmic form in order to define and develop the main articulations of the ICA-3M. Generally, the meeting follows a model of interactive dialogue by regulated exchange of voice messages between participants alternately according to a principle of distribution of the right to speak. However, only the nature of the voice message exchanged is important in determining the useful part of the meeting of the part considered to be noise. For algorithmic modeling, we consider that the process is held according to the following list of successive actions or steps:

1. Open the meeting
2. First welcome speech

3. Register the Present on the attendance list
4. Announce the plan: a list of points or problems noted PTk to be addressed
5. Validate the plan: change the order of the points, introduce new points
6. Announce the plan-related PTk issue with the gain G to be won
7. Open and collect SLq entries for the plan-related PTK issue

8. Distribute the TSA
9. Get actors' contributions
10. Check if there are any new contributions on PTk, if so, increment q and go to 7
11. Summarize the proposals for the PTk problem and mark those selected
12. Assign the G win to each author of a successful proposal
13. If the agenda is not exhausted, increment k, move on to the next issue PTk, go to 6
14. Summarize the proposals selected for each problem
15. Calculate and elect the Best Contributor Actor
16. Record the end time
17. Prepare a report of the meeting

**Format Conversion Method for ICA-3M**

**Basic principle**

Whether it is a question of its consideration in the form of audio or video recordings for an observer, a meeting is first presented in a continuous format. As a result, we define here the principle to be used for the conversion, knowing that it is the discrete audio captured in useful sequences that will be converted into structured text format by the transcription algorithms. These algorithms allow to switch from audio words spoken in actors' speeches to their equivalent in text format. Thus, we define the following parameters:

- ✓ Meetings are identified by numbers from 1 to $NBR_{max}$ (or MAXimum Number)
- ✓ $NBPR_{MAX}$ is considered to be the MAXimum Number of Properties of a Meeting. This may include, for example, the place, the reasons for the summons, whether ordinary or extraordinary, the date and time of the start and end that we will record, etc.
- ✓ The actors are identified by numbers that range from 1 to $NB_{ACT}$ (Number of ACTors)
- ✓ $NBPA_{MAX}$ is considered to be the MAXimum Number of Actor Properties. For example, these can be surnames, first names and position as the first properties that we will record, accompanied by age, gender, date of birth, place of birth, nationality, ...
- ✓ Problems (PT) or items or issues on the agenda of a meeting are identified by increasing numbers that evolve from 1 to $NB_{PT}$ (Number of Problems Addressed)
- ✓ Meeting Proposals (PR) identifiers range from 1 to $NBG_{PR}$ (Global Number of PR)
- ✓ Speaking Lists (SL) identifiers range from 1 to $NBSL_{max}$ (Maximum Number of SL)
- ✓ One problem can lead to the opening of several SL

The departure agreements are stated : meetings are held several times a year, whether ordinary or extraordinary. During a meeting, the first actor to state a proposal becomes the author. We consider a memory representation of C language arrays whose indices start from 0. The transition to structured mode is done through the AT-SAM/DS algorithm presented below. It requires defining the following variables and symbolic constants:

- ✓ Global List of Actors (GLA): ($NB_{ACT}$*$NBPA_{MAX}$) dimension matrix type. Its line indices correspond to the actor identifiers by adding 1. Its column 1 contains the first and last names of the actors. The column 2 is for functions of the actors, other properties of an actor are ignored (located in database tables).
- ✓ Global Meetings List (GML): ($NBPR_{MAX}$ * 3) dimension matrix type. Its line indices correspond to meeting numbers by adding 1. The column 1 is for dates of the meetings, column 2, for meeting start times, column 3, end times of the meetings.
- ✓ Global List of All Problems Addressed ($LG_{PT}$) or items on the agenda of a meeting: (NBPT * NBRmax) dimension matrix type. Its line indices represent problem numbers by adding 1. The column indices correspond to the meeting numbers by adding 1. The boxes are of the composite structure type that contain the label of each problem associated with the gain attributed to the best proposal on that problem.
- ✓ Global List of All Meeting Proposals (GLMP): ($NB_{GPR}$ * 4) dimension matrix type. It establishes an equivalence between a proposition contained in a box (i) with its number represented by i the line index of this box. Its line indices represent the numbers of the proposals by adding 1. The column 1 is for wording of the proposals in strings, column 2, number of the problem being addressed, column 3, author's identifier, column 4, G if the proposal was accepted and 0 if not.
- ✓ Attendance List (AL): ($NB_{ACT}$ * NBR max) dimension matrix. Its line indices represent the number of the actors by adding 1. The column indices correspond to the meeting numbers by adding 1.
- ✓ Speaking List (SL): matrix of dimension ($NB_{ACT}$ * $NBSL_{MAX}$). The indices of the lines represent the number of the actors by adding 1. The indices in the columns are numbers of the open SL for all the issues discussed in a meeting, adding 1

In the table below, the variables or parameters i, j, k, p, q are index of the lists. They are therefore replaced by their numerical values for oral pronunciation: i is used for meetings in the range of 1 to $NBR_{max}$. The j index is used for the actors' registration numbers (from 1 to $NB_{ACT}$). The k index is used for Processed Problems (from 1 to $NB_{PT}$). The q index is used for Speaking Lists (from 1 to $NBSL_{MAX}$). The p-index is used for meeting proposals (from 1 to $NB_{GPR}$).

The following table represents our sampling grid for the transition from continuous meeting mode to structured mode with useful text recorded in the previously defined tables that represent the ICA-3M data structure. It summarizes the conventional keywords that are the parameters of the AT-SAM/DS algorithm for switching to discrete format from the continuous format of the normal meeting flow. As a seamless transition, we use the transcription of the discreet useful audio format of a meeting to the text format, to fill in the DS and then the tables of the database and finally generate an automatic final report by offering the possibility of later translating this report into another language.

| Keywords spoken by the actors | Equivalent or synonymous keywords | Corresponding algorithmic action or instruction |
|---|---|---|
| We open the meeting i on this day dd/mm/yyyy from the start time H and M mn | Welcome to the meeting i on this D-day from the hour H and M minutes | Enter the date dd/mm/yyyy in GML (i, 0)<br>Enter H:M in GML (i, 1) |
| We open registrations on the attendance list of the meeting i | You can sign up for the meeting i attendance list | Initialize AL Boxes to 0<br>By default, no actor is present |
| Sign me up, I'm Mr. j | Mr. j you are registered | For any actor j present, write 1 in the AL box (j, i) |
| The agenda is as follows:<br>Problem number k<br>Reads as follows: «Wrd»<br>With G gain | Here is the agenda:<br>issue number k «Wrd» with the gain G / Point k «Wrd» with the gain G | For any problem k<br>Write «Wrd» in<br>LGPT (k, i).Wording<br>Write G in<br>LGPT (k, i).Gain |
| We will move on to the issue number k on the agenda | We are addressing problem k of the agenda<br>Agenda issue k | Save k in a VT0 working variable |
| We open the q speaking list for problem k | You can sign up for the q speaking list of k issue | Save q in the working variable VT1 |
| Mr. j you are registered | Dear Mr. J | Write 1 in the box<br>SL (j, VT1) |
| We close the speaking list q for problem k | If there are no other volunteers, we close the SL q of the $PT_k$<br>End of registrations on | Initialize the p-counter of the propositions to 1 |

| | | |
|---|---|---|
| | the SL q of PT $_k$ | |
| Mr. J, you have the floor | I give the floor to Mr. J<br>It is up to you, Mr. J<br>It's the turn of Mr. J | Save j in a VT2 working variable |
| I propose that for problem k: "text of proposal p" | For problem k, I propose:<br>For problem k, we propose:<br>We propose for problem k: | Write "text of proposal p" in the GLMP (p, 0), Write k in GLMP (p, 1), Write VT2 in GLMP (p, 2), Write 0 in GLMP (p, 3)<br>Increment p |
| Mr. J, you have been registered | Mr. j has been registered | Write 0 in the box SL (d, q) |
| We close the speaking list q for problem k | If there are no other volunteers, we close the SL q of the $PT_k$<br>End of the registration on the SL q of the $PT_k$ | Save the value of the p-counter in the Working Variable VT3 |
| Here are all the proposals for $PT_k$: "text of the p proposals " | We give the list of proposals for $PT_k$ | For p ranging from 1 to VT3 display GLMP (p, 0), GLMP (p, 1), GLMP (p, 2) |
| Here are all the proposals p retained for $PT_k$ with the gain G | For $PT_k$, we retain the propositions p/We assign the gain G to the authors of the proposals p | For each value of p, Write G in the GLMP box (p, 3) |
| Are there still reactions to the problem k | We are opening a final SL q for $PT_k$ | Save q in the working variable VT1 |
| We close the meeting with the following summary: "*reading the text of the automatic report*" | We have now come to the end of the meeting. I will read you the summary: "*reading the text of the automatic report*" | View attendance list, Agenda<br>For each agenda item, display the list of proposals, indicate those selected with their authors, Calculate, display winnings, Electing the BCAM |
| End time is H:M | We close the meeting at H:M | Enter in GML (i, 2) the time H:M |

Table 1: Equivalence between Keywords and Algorithmic Actions (3AKT)
Legend: In red are the keywords and in yellow the significant changing values

**AT-SAM/DS Algorithm**

It presents the course of a meeting with the actions in the 3AKT, the transcription of the audio to the text is transparent. That text will be recorded in the ICA-3M DS.

1. Open the meeting
   Calculate the order number i of the meeting from the uploaded data
   Enter the date dd/mm/yyyy in GML (i, 0), Enter in GML (i, 1) the time H:M
2. First welcome speech

3. Register the Present on the attendance list
   Initialize AL Boxes to 0
   For any actor j present, write 1 in the AL box (j, i)
4. Announce the plan: a list of points or problems noted $PT_i$ to be addressed
5. Validate the plan: change the order of the points, introduce new points
   Initialize k to 1
   For any problem k stated
   Write «Wrd» in LGPT (k-1, i).Wording, Write G in LGPT (k-1, i).Gain
   Reset k to 1 for the incoming instructions
6. Announce the plan-related $PT_k$ issue with the gain G to be won
   While the agenda is not exhausted
   Save k in the working variable VT0, Initialize q to 1, Initialize p to 1
7. Open and collect SL q entries for the plan-related $PT_K$ issue
   While there are contributions on $PT_K$
   Save q in the working variable VT1
   For any number j of an actor stated Write 1 in SL (d-1, VT1-1)
8. Distribute the lyrics (order of registration on the SL)
   For j ranging from 0 to (NBACT – 1)
   If SL (j, VT1-1) = 1 then
   Giving the floor to the actor j
   Save j in VT2 (statement not needed, just don't close the "if" to include the following instruction "9.")
9. Gathering input from stakeholders
   Write "text of proposal p" in the GLMP box (p-1, 0) Write k in the box GLMP (p-1, 1)
   Write VT2 in the box GLMP (p-1, 2)
   Write 0 in the box GLMP (p-1, 3), Increment p
   End if
   End for
10. Check if there are any new contributions on $PT_k$, if so, increment q and go to 7
    Increment q
    End while
    Save the value of the p-counter in the Working Variable VT3
11. Summarize the proposals for the PT k problem and mark those selected
    For p ranging from 1 to VT3
    write GLMP (p, 0), GLMP (p, 1), GLMP (p, 2)
12. Assign the G win to each author of a successful proposal
    If p is retained, write G in GLMP (p, 3) end if End for
13. If the agenda is not exhausted, increment k, move on to the next issue $PT_k$, go to 6
    End while
14. Summarize the proposals selected for each problem
    View attendance list, View Agenda, For each item on the agenda, display the list of proposals and indicate those selected with their authors
15. Calculate and elect the Best Contributor
    Calculate and display winnings, Electing the Best Contributor

16. Record the end time

    Enter in GML (i, 2) the time H:M

17. Prepare a report of the meeting

    Write the following information in the "REPORT" file:

    The date and time of the meeting, The attendance list, The agenda

    For each agenda item,

    Write the list of proposals with an indication for each selected proposal of the name and surname of its author and its prize

    The name and surname of the actor elected Best Contributor

    The end time of the meeting

**Contribution of the ASR to the automation of a meeting**

In the principle of the meeting, the chairman of the session authorises each participant registered on the SL in turn to take the floor and to verbally state his contribution or proposal, generally with an argument. Automating this step is possible with the ASR or Automatic Speaker Recognition. The aim is to allow this system to detect the voice of each actor who has spoken to provide the ICA-3M with his number. To do this, this algorithm just needs to read from the GLA table, which will be improved in this context with an additional column containing the voice samples of each actor. After the speaker has been recognized, the audio-to-text converter is then invoked to produce text that will be placed at the entrance of the NAL in order to separate the proposal from the developed argument. Thus, the ICA-3M retrieves the actor's number and will then be able to enter the text of the proposal in the GLMP table. One of the contributions of the ASR also lies in the possibility of detecting cases of spoofing by violation of the SL order or even without registration on the SL. To do this, the ICA-3M will function as a state machine by constantly checking whether the order of the authorized sequences is not compromised as the actors' speeches are captured and sentences are formed.

**Contribution of NAL to the automation of a meeting**

During a meeting, for the recording of proposals by actor, separating them from the argument he has developed, we plan to automate this step with the system NAL or Natural Language Processing, in particular its component of the ATST or automatic Text Summary Technic. Like what following the transcription of the audio to the text and after the separation phase between the proposal and the argument, it is also useful to provide a qualitative and quantitative assessment of each proposal, to automatically judge its proximity to the solution to the problem or agenda item at hand. This automatic judgment can only be developed from a learning corpus or by analyzing the reports of the meetings. In this case, we need to define the notion of distance between the proposal and the solution, with the possibility of integrating the constraint imposed by the chairman by setting out the criteria that any proposal must verify.

**Architecture of the new system with ASR and NAL**

**System architecture**

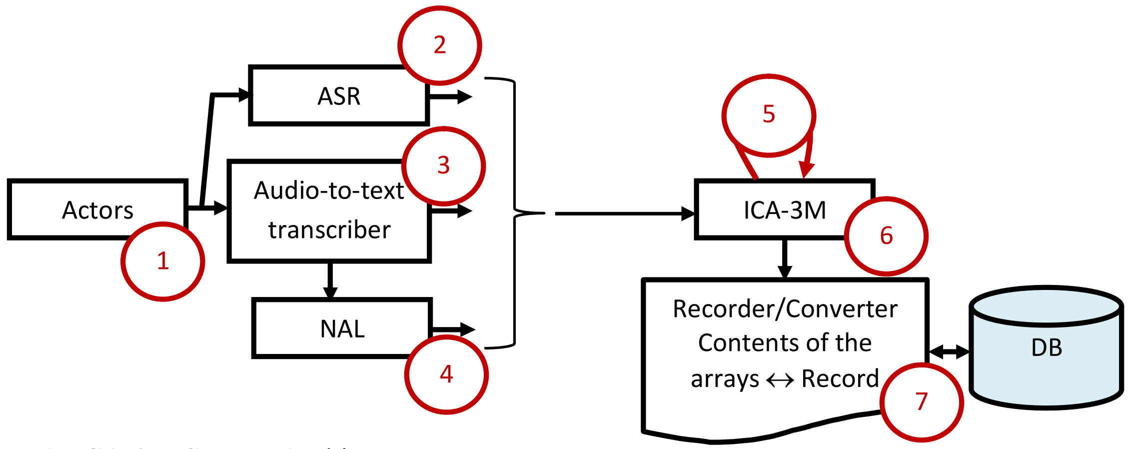


Figure 1: ICA-3M System Architecture
Legend:

- A meeting participant speaks
- The ASR system identifies it and gives its number to the ICA-3M
- The transcriber transforms speech into text and transmits it to the ICA-3M and the NAL
- The NAL separates the proposal from the argument and forwards it to the ICA-3M
- ICA-3M Updates Internal Tables
- The ICA-3M transmits to the recorder the information to be stored in the database
- The recorder updates the database.

## Conclusion

In this paper, we describe the meeting basic principles and determined the contribution of VoIP to the success of current video conferencing platforms. It is a technology based on protocols including the H.323 family and the SIP family. We highlighted the limitations of these platforms before proposing our innovative approach based on the new ICA-3M system. It uses principles that include format conversion techniques to evolve from the audio and video recordings of the meeting to a text format with a sampling method. We proceeded to the algorithmic modeling of the course of a meeting and proposed the AT-SAM/DS algorithm for the format conversion in the ICA-3M system. The strength of this algorithm lies in the use of specific data structures to switch from audio/video format to text format, which offers, among other advantages, that of automatically generating a meeting report. Similarly in this paper, we determined the contribution of ASR and NAL to the automation of a meeting, which allowed us to propose an architecture of the new system integrating these two booming fields.

As a future work, in the structures of the ICA-3M system, we have planned fields to determine the Best Contributor. To do this, we plan a mathematical modeling before the evaluation of the performance of the overall system by simulations with, among other criterias, the measurement of the error rate and the success rate in the format conversion. Similarly, the integration of the possibilities offered by the

Internet of Things (IoT) will offer a new channel that will allow the extension and inclusion of remote and meeting room outside actors.

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