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"Network Music Performance – Problems, Approaches and Perspectives"

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Abstract

With the current Internet bandwidth capacities and machine processing performance the personal computer has become an affordable and flexible multimedia platform for high quality audio and video content. Besides the delivery of services such as TV, telephone and radio, the Internet can also be used for the exchange of musical information. Due to the variety and complexity of already existing remote music and communication approaches, an elaboration on this topic is mandatory, which covers any relevant musical, technical or interdisciplinary aspect of remote musical interaction. Therefore, this paper gives an overview of current applied technologies and possibilities with their theoretical background.

1 Introduction

Originally the term "Network Music Performance" was initiated by John Lazzaro from Berkeley University, in 2001 [1] and since then it has mainly been used for the description of distant musical interaction on the Internet. Within the current emancipation and commercialization of this topic other terms like DIP (distributed immersive performance) [2], eJamming [3], NinJam [4], Quintet.net [5], SoundWIRE [6] or Soundjack [7] have come up. Though each of them follows its own approach, in fact all of them try to describe the same thing: musical interaction on distance using the Internet [8].

In order to realize distant interaction, data has to be transmitted from a sender to a receiver, similar to the telephone or radio principle. This transmission implies a certain delay. Since music is a very time sensitive form of communication, this delay should be as short as possible [8]. Unfortunately and due to its technical characteristics, the Internet cannot be considered as an ideal medium for the transmission of short delayed audio data and hence certain technical or musical compromises have to be taken into account [8]. In consequence, Network Music Performance could not gain wide acceptance in daily practice, – yet especially among professional musicians. However, most musicians rate such kind of technology generally as useful.

In this paper we will present a classification of existing sate-of-the-art solutions for Network Music Performance. Instead of just focusing technical aspects of the different approaches we will also present an in-depth analysis of the musical aspects that are crucial for using this kind of technology for rehearsing and performing music in its natural way.

This paper is structured as follows: In section 2 we will categorize different musical interaction styles. As we will see here, there is a great variety of styles and each one implicates an individual tolerance to delay. In section 3 we will identify potential sources of delay caused by electronic equipment and the Internet. In section 4, we will present some typical delay boundaries of Internet connections in order to give the reader some guidelines which of the previously presented interaction styles are feasible in a certain use case and which are not.

2 Categorization

Generally, in rhythmical music we have to distinguish between a solo instrument and a rhythm instrument. Apart from that the placement of the so-called “rhythm section” is of significant importance. As an example with drums, bass and saxophone, a simple case is present in which drums and bass form the “rhythm section” while the saxophone player represents the “solo section”. Though of course the solo section and any musician must have a sense of rhythm, it is basically the interplay of bass und drums which forms the essential fundamental groove of an ensemble which allows other solo instruments to play upon. In this scenario the saxophone player relies on and plays on the groove that is produced by the rhythm section [8]. Due to the fact that rhythm and synchrony are the main fundament of groove based music, the following sections put emphasis on rhythm based instruments and the groove building process.

In classical music things are more complex: Here we usually cannot precisely distinguish between “rhythm” and “solo” sections. Anyhow, in most pieces of classical music an analogical categorization is feasible, but should be more fine-grained and more dynamic. Also the concept of a conductor has to be considered here. In the following – in order to present our concepts as clear as possible to the reader – we will focus on applications in the field of rhythmical music and continuously use the according terms “solo section” and “rhythm section”.

Taking the signal delay between two musicians as the critical factor and increasing it from zero to infinity, we can separate the possibilities of a musical interplay into four main categories A to D. Category B can be divided into three subcategories B1 to B3.

2.1 Category A - Realistic Interaction Approach (RIA)

A realistic musical interaction, as if in the same room, assumes a stable one-way latency of less than 25 ms [8][9] between two rhythm-based instruments such as drums and bass. In this scenario both instrument’s grooves merge into each other and the real musical interplay can happen [9]. From the perceptual point of view the delay appears to be as not existing, which is similar to musicians playing with a maximal physical distance of eight meters in a rehearsing space, where the speed of sound is the limiting time delay factor. The RIA is the only approach professional musicians accept without any compromise since it is the only scenario, which exactly represents the conventional process of creating music in groups or bands [10]. Beyond this threshold of 25 ms, the groove-building-process cannot be realized by musicians anymore and thus different compromises and categories have to be applied.

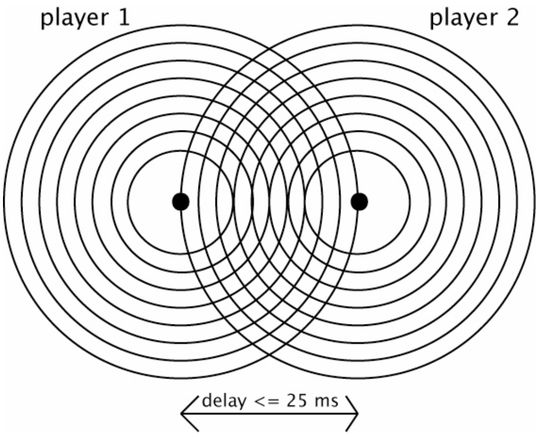


Fig. 1: Two players in RIA mode

Due to technical difficulties in applying the required RIA conditions, RIA has so far not turned into a commercial entity but has mainly been examined in research projects, such as SoundWIRE by Chris Chafe of CCRMA and our Soundjack system.

2.2 Category B1 – Master Slave Approach (MSA)

Assuming an attendance to compromise and to step back from musical perfection and ideals, it really is feasible to perform with two rhythm-based instruments such as drums and bass, even when exceeding the 25 ms threshold – simply if one of the musicians keeps track of his rhythm and does not listen to the incoming high delayed signal anymore. In that situation the remote side can perfectly play to the incoming signal since the other side doesn't care about the response anymore – a change in the musical interaction is happening, which here is called the "Master-Slave"-Approach. The first musician takes the master role since he is producing the basic groove while the remote musician simply relies on it and hence takes the slave role [9]. Of course the higher the delay, the more difficult the ignorance of the delayed input can be realized by the master since shorter delays will easier establish a musical connection to the previously played notes. In terms of delay MSA generates no latency and perfect sync on the slave's side but on the other hand it delays the slave with the roundtrip delay on the master's side. While the slave musically depends on the master but has a perfect sync, the master has musical independency but an unsatisfying sync.

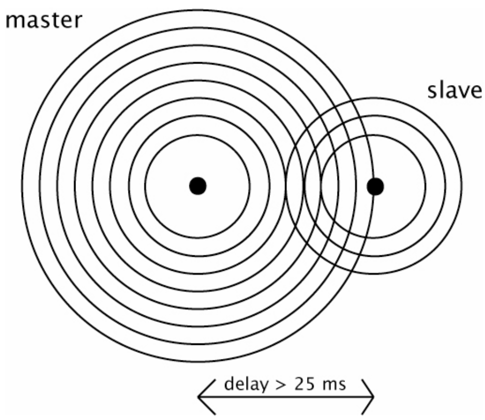


Fig. 2: Two players in MSA mode

In general the master role is taken by a rhythmic instrument in order to let solo instruments play on its groove in slave mode. An exception can happen when a rhythmic instrument suddenly starts with a solo part. In this case it will require the other instrument to take over the leading rhythmic role, which in turn leads to a switch of roles.

MSA can be applied with any system that allows the transmission of realtime data on the Internet. This could be tools for IP telephony or videoconferencing, which do not put emphasis on low delay signal transmission, but as well high speed audio transmitters in an intercontinental setup. In the latter case the main source of latency is the long physical distance.

2.3 Category B2 – Laid Back Approach (LBA)

The Laid-Back-Approach is based on the “laid back” playing manner, which is a common and accepted solo style in jazz music. Playing “laid back” means to play slightly behind the groove, which musicians often try to achieve consciously in order to make their solo appear more interesting and free.

The Laid-Back-Approach is similar to the Master-Slave-Approach and is mainly determined by the number of participating instruments and their role. As previously mentioned, two rhythm-based instruments separated by delays beyond 25 ms have to play with MSA but in case one of the instruments being a solo instrument, the situation changes.

Exchanging the drums with a saxophone in the example scenario results in a remote rhythm/solo – constellation in which the bass represents the rhythm instrument and the saxophone the solo instrument. Since the bass now has no rhythmic counterpart anymore, it alone takes the responsibility for the groove while the saxophone plays its solo part on it. Equal to MSA to saxophone has a perfect sync on its side and is transmitted back with the roundtrip time but in comparison to MSA this has no disturbing effect for the rhythm instrument in LBA. The saxophone is delayed by the roundtrip delay time, which adds an artificial laid back style on it and hence this playing constellation is not to be considered as problematic anymore.

LBA of course doesn’t work for unison music parts in which both parties have to play exactly on the same beat at the same time. Beside this, “laid back” implies that conscious or artificial delay must not exceed certain limits. Beyond these limits the incoming signal has no context to the previously played musical fundament, and at that point the term “laid back” switches to the term “out of time” which will be described within the following LAA (Latency Accepting Approach) paragraph. The additional delay on the master’s end ranges between 50 ms up to a maximum of 100 ms but still depends on the musician’s subjective perception and the bpm (beats per minute) of the actual song.

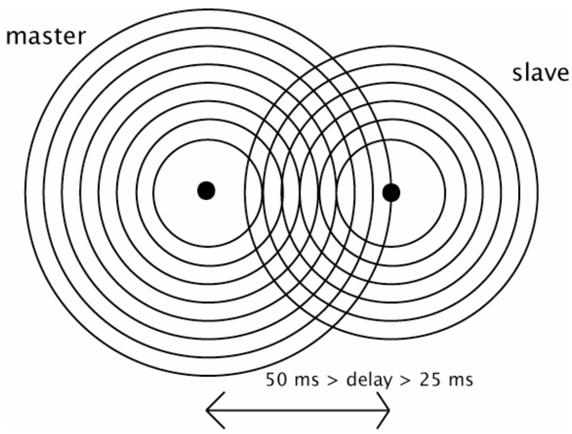


Fig. 3: Two players in LBA mode

With more than two instruments the actual instrument constellation is of significant importance. In the bass/drums/sax music scenario in two locations, separated by a 20 ms delay, there can be three constellations:

In constellation A the rhythm section plays together on one side and the saxophone player is placed on the remote end. The rhythm section will play a solid groove on one end, which is transmitted to the saxophone player. Since the saxophone player represents a solo instrument, he is not part of the groove building process, thus will simply use the incoming solid music as a basis for his solo play. This in turn means that he will receive the groove with a 20 ms delay but his solo play again takes 20 ms to be transmitted back to the rhythm section, resulting in a 40 ms delay for the rhythm section. Hence, even the one-way delay between the two sides is below 25 ms, the musicians switch from RIA to LBA. In example scenario B and C the rhythm section will be separated in two places and since the groove is produced synchronized on both ends, the saxophone's location is not an issue anymore. Here RIA is applied in contradiction to scenario A.

LBA is used when the delay ranges in areas slightly beyond the 25 ms RIA threshold. Again SoundWIRE and Soundjack represent potential candidates, beside the Musigy [14] software as one of the fist commercial products. It provides audio delays, which range at the edge between RIA and LBA.

2.4 Category B3 – Delayed Feedback Approach (DFA)

In case the 25 ms delay threshold is exceeded, DFA tries to make musicians feel like playing with the RIA by delaying player's own signal artificially: By principle delays beyond 25 ms lead to either LBA or MSA styles in which the master hears the slave with a delay equal to the roundtrip time while the slave plays in perfect sync. When delaying the playback of the master's signal, both sounds finally have a closer proximity at the master's ear, which improves the problematic delay gap in MSA or reduces the laid back effect in LBA. The larger the self-delay the better the synchronization of both signals. The best synchronization can be reached with a self-delay equal to the roundtrip-time.

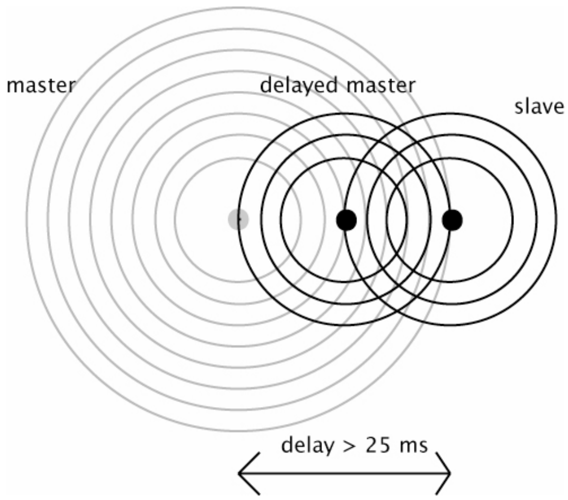


Fig. 4: Master player delayed in DFA mode

Anyhow, we have to mention that a spontaneous switch in the master / slave's role will lead to a worse situation for both players than it would appear without an artificial delay. In this particular scenario the none-delayed master performs like in a normal not delayed MSA or LBK scenario but the now delayed slave will not to be able to play in sync with its master track due to a possible confusion by its own delay.

A second variation of DFA is present when both instruments are delayed by the one-way delay and a metronome sounds in both places. Assuming the metronome sounds at precisely the same time, both instruments hit the beat and their sound is transmitted to the destination with the one-way delay. Since each signal is delayed by exactly the same latency, both sounds will appear at the same time in both locations, no matter how large the delay is. Though this principle can effectively improve the delay situation between two players, the user has to pay attention with the precise adjustment of the artificial self-delay: With values beyond the one-way latency, the remote signal will sound earlier than the local signal and hence leads to unnatural delay shifts between the two players.

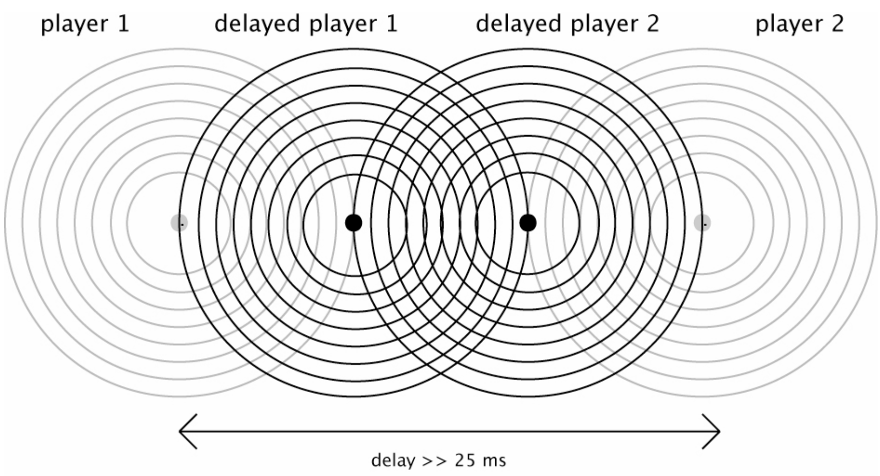


Fig. 5: Both sound sources delayed in metronome triggered DFA mode

Though DFA improves the delay situation between two musicians, it is no doubt that a delay of one's own signal typically can be considered as inconvenient and not natural. The larger the delay gets and the louder the instrument's direct noise, the worse the realistic instrument feel and playing conditions. This is especially valid for any acoustic instrument such as a violin or drums. On the other hand DFA can be a suitable approach for the synchronization of remote play back sound sources. In case i.e. two DJ's turntables are connected with each other, a delay of the turntable's output would not lead to timing-problems. A machine's play back behavior does not depend on an inner time or feel like human beings do and can of course reproduce delayed sounds without loosing any kind of rhythm.

The most famous DFA software is the eJamming system, which automatically adjusts the relevant delay parameters and has so far been the most famous and best-known commercial product for music on the Internet. Besides this, a research group at the University of Braunschweig,Germany has been actively developing and presenting a software solution based on DFA [14].

2.5 Category C - Latency Accepting Approach (LAA)

While all previous approaches try to find alternative ways for realistic network music performances, the latency accepting approach steps back from latency optimized or compromised solutions and simply accepts delays beyond 25 ms. In principle LAA has no motivation to create conventional music and thus can allow any delay which is consciously taken it into account. In this scenario musicians play with the delay and use it as an artistic way of expression. LAA is the most avantgardistic approach resulting in a total dissociation of musical conventions and function with the Internet as the core technology.

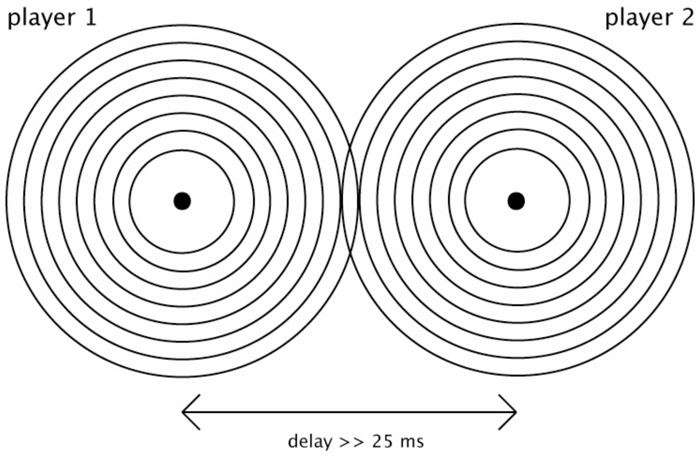


Fig. 6: Rhythmically unrelated sound sources in LAA mode

In terms of new avantgardistic music in LAA, the Quintet.net framework by Georg Haydu fulfills relevant requirements and can be applied under any kind of network condition. Quintet.net transmits MIDI control data and does not necessarily require the user to play a musical instrument. The user can play with an electronic input device for the sound generation instead. Apart from that, various worldwide network sessions with SoundWIRE have taken place, in which modern, new music is the dominating style of performance.

2.6 Category D – Fake Time Approach (FTA)

As an alternative to the previous approaches the FTA (Fake Time Approach) [5] puts emphasis on musical experimentation and expression and tries to establish a jamming environment under the assumption that network latency prevents true real-time synchronization of the participating musicians. To circumvent this assumption, a communication principle is introduced in which latency is increased in such a way that participating performers receive each other's output with the delay of exactly one measure. In fact musicians play asynchronously to the music their colleagues have played one measure before. Of course this principle doesn't really allow a realistic interplay but keeps up the vision of such and has a practical identification: In many so called rock or funk music jam sessions the same basic groove is often played over and over again without any explicit difference to previous measures. In this case FTA might satisfy the musician's needs more than any previous approach apart from RIA as the ideal case scenario. In terms of delay improvement FTA is in so far the most interesting approach, since it increases the delay up to one measure instead of reducing it as all other approaches do.

The only representative for the FTA is the Ninjam-Project. Ninjam is a successful client / server based open source software which delays any music stream as described and hence can be applied in any kind of network scenario. Due to this versatility and flexibility Ninjam has been used in virtual environments such as second life and can meanwhile be considered as a competitor to the commercial eJamming software.

2.7 Category Characteristics

The choice of a category has an influence on the player's perception. In this context five musical aspects matter significantly. In the following table "Sender Sync" and "Receiver Sync" describe the rhythmical synchronisation of the players' signals in a musical interaction. "Own Feel" is an indicator for how realistic playing the own instrument feels for each player. "General feel" generally reflects, how realistic the session feels in comparison to a conventional music session in the same room, while "Unisono play" shows, if musicians will be able to play the same notes at the same time. How well an aspect is fulfilled by each category, is indicated by "+" (good), "o" (average) or "–" (bad).

	Sender Sync	Receiver Sync	Own Feel	General feel	Unisono play
RJA	+	+	+	+	+
MSA	–	+	+	o	–
LBA	o	+	+	+	o
DFA	o	o	o	o	o
LAA	–	–	+	+	–
FTA	o	o	+	–	–

Table 1: Summary of play characteristics of each approach

3 Key factors

The key factors in audio processing for Network Music Performances are signal quality and signal delay. It is the available bandwidth, which determines the quality of an audio signal. An uncompressed full quality audio stream of one channel requires 768 kbps plus additional bandwidth produced by the Internet protocol layers ranging from 50 to 300 kpbs, which results in a maximal bandwidth of 1 Mbps. For local area networks with capacities larger 10 Mbps up to Gigabit connections this amount of traffic can be considered as relatively low but in home DSL lines with upload sizes typically less than 1 Mbps the stream cannot be sent anymore. Hence if bandwidth is a rare resource, the amount of data has to be reduced somehow. On one hand compression codecs can be used, or on the other hand the audio data can be exchanged by MIDI (Musical Instruments Digital Interface) or control data. While the latter produce a very low amount of network traffic of some kilobytes, the former results in a tradeoff between compression efficiency and processing delay.

Assuming sufficient network bandwidth resources and/or a suitable signal size, it is mainly the physical distance and the soundcard configuration, which determine the total audio latency. This interrelation is described in the following paragraphs.

3.1 Network Delay

Electrical signals travel with approximately 70 % of the speed of light, which equals a speed of ~ 210.000 km/s or a delay of 5 ms for a distance of 1000 km and in fact it is mainly the physical distance between two locations, which determines the delay. Unfortunately, direct links between sender and destination do rarely exist and hence the actual route depends on the structure of the used Internet backbone [12].

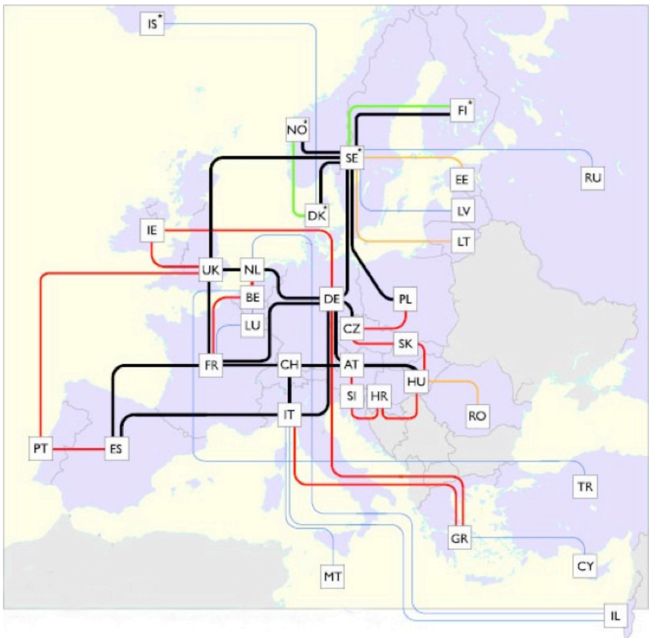


Figure 7: Continental Routing Scenario (GEANT backbone structure [11])

A good example for an Internet backbone is the GEANT backbone (see Figure 1), which interconnects various research facilities all over Europe. It is clearly visible that in general direct paths do not exist. I.e. a sender in northern Germany can reach a Danish destination only via detours to south Germany and Sweden, which in turn leads to higher delay times.

In numerous practical tests we observed, that the delay threshold of 25 ms can be kept up to a distance of approximately 1000 km. This rule of thumb is demonstrated in a practical example. The University of Lübeck in Germany and SARC (Sonic Arts Research Center) Belfast in Northern Island are separated by a direct path of 1045 km but due to a significantly longer network route the roundtrip time ranges at 54 ms, which equals a one-way delay of 27 ms.

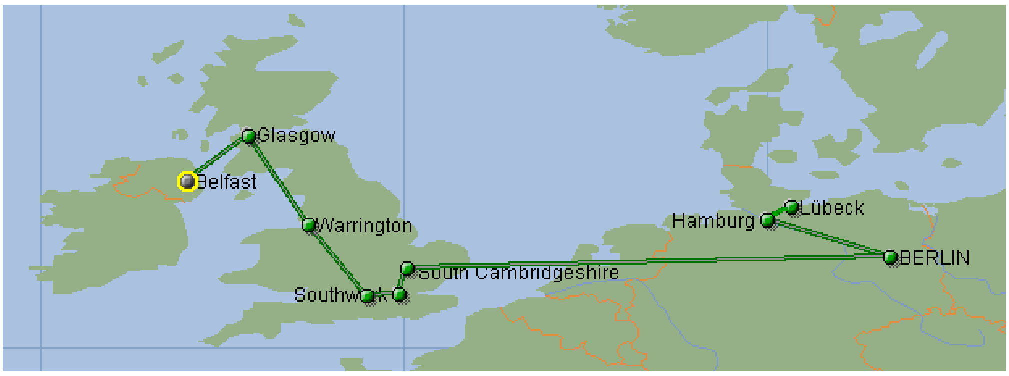


Figure 8: Route between Lübeck and Belfast

In comparison to national or continental connections, intercontinental connections typically range in way higher delay dimensions. For the following example route between the University of Lübeck,Germany and CCRMA (Center for Computer Research in Music and Acoustics) Standford,USA (Figure 9), we measured a roundtrip delay of more than 200 ms. The graph in Figure 10 shows, which hops the signal passes on the way from the sender to the destination and how large the delay for each hop is. The transatlantic signal delay jump of more than 110 ms (roundtrip delay) between hop 11 and 12 gives a clear view of the latency/distance dependency. Due to common transmission irregularities in IP networks, some hops respond earlier or later than expected. With respect to the next chapter, these delay variations represent a further source of problems.

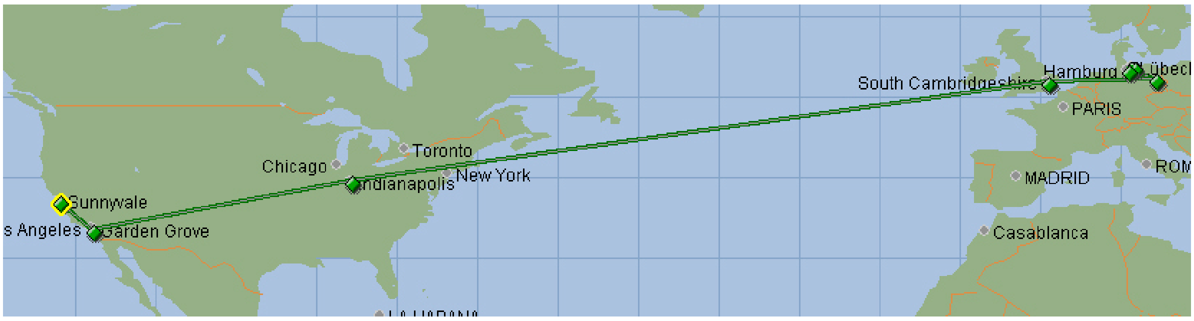
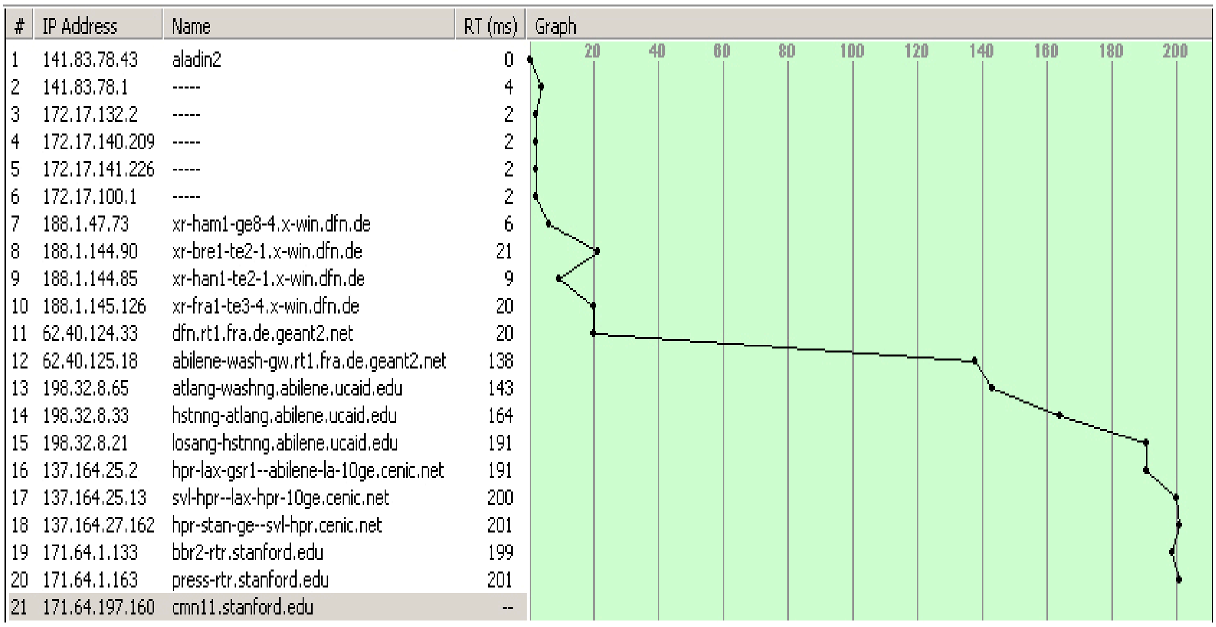


Figure 9: Route between Lübeck and California, USA



#	IP Address	Name	RT (ms)
1	141.83.78.43	aladin2	0
2	141.83.78.1	-----	4
3	172.17.132.2	-----	2
4	172.17.140.209	-----	2
5	172.17.141.226	-----	2
6	172.17.100.1	-----	2
7	188.1.47.73	xr-ham1-ge8-4.x-win.dfn.de	6
8	188.1.144.90	xr-bre1-te2-1.x-win.dfn.de	21
9	188.1.144.85	xr-han1-te2-1.x-win.dfn.de	9
10	188.1.145.126	xr-fra1-te3-4.x-win.dfn.de	20
11	62.40.124.33	dfn.rt1.fra.de.geant2.net	20
12	62.40.125.18	abilene-wash-gw.rt1.fra.de.geant2.net	138
13	198.32.8.65	atlang-washng.abilene.ucaid.edu	143
14	198.32.8.33	hstnng-atlang.abilene.ucaid.edu	164
15	198.32.8.21	losang-hstnng.abilene.ucaid.edu	191
16	137.164.25.2	hpr-lax-gsr1--abilene-la-10ge.cenic.net	191
17	137.164.25.13	svl-hpr--lax-hpr-10ge.cenic.net	200
18	137.164.27.162	hpr-stan-ge--svl-hpr.cenic.net	201
19	171.64.1.133	bbr2-rtr.stanford.edu	199
20	171.64.1.163	press-rtr.stanford.edu	201
21	171.64.197.160	cmn11.stanford.edu	--

Figure 10: Traceroute between Lübeck and California, USA

3.2 Soundcard Delay

Digital soundcards process their data in a blocking manner, which means that the device blocks for an amount of time in order to read a specified number of samples (which make up a so called "frame") [12]. Hence an analog signal is captured by first generating a frame of a fix number of samples, which now can be modified digitally by the user and played out afterwards. This analog/digital and digital/analog conversion happens frame by frame and leads to a continuous audio stream. The blocking time for the processing of audio data appears on the capture side and the playback side is determined by the frame-size, which can be specified by the user. Unfortunately, the lowest frame-size is determined by a combination of system performance, operating system and soundcard drivers. In general 128 samples per frame is the lowest limit [12].

In terms of sending audio frames across a network, one remark is of high significance. The propagation time for packets on the Internet might not be of reliable constancy. Depending on the network structure and conditions, the sent packets typically suffer from a delay variation, which appears with a certain probability. This delay variation is typically called the 'network jitter'. Too late arriving packets prevent the receiving soundcard to play out a solid audio stream and produce disturbing cracks instead. In a situation, in which the user wants to reduce audio dropouts, two solutions can be applied: On one hand the user can increase the audio frame-size. The larger the audio frame, the less frames have to be sent in the given time range and the less probability for a frame to be affected by a delay variation. Another solution is an increase of the network audio buffer in order to store a higher amount of packets. Once a certain buffer-size is reached, the soundcard will start reading from it. In case a packet arrives too late, the buffer will still have older packets in the queue and no crack will appear.

Both solutions provide a higher play-out reliability, which in turn increases the play-out latency. This trade-off is the most significant aspect of Network Music Performances and the user is forced to carefully deal with it. The consequence is that the more stable a network is, the lower the audio delay can be adjusted.

4 Latency Boundaries for Practical Applications

As the conclusion of the previous chapters, it is the physical distance and the network stability, which form the fundamental conditions of a network music performance. In order to make a common statement for any possible network case, we generalize both parameters into groups of three. The physical distance is divided into short (1 km - 1000 km), medium (1000 km - 3000 km) and long (> 3000 km). The stability is classified into constant (0 ms - 2 ms jitter), medium (2 ms - 20 ms jitter) and instability (> 20 ms jitter). The combination of both parameters determines the total audio latency, which determines possible application categories (presented in section 2). Depending on this actual combination, Figure 11 shows, under which conditions specific NMP categories can be applied.

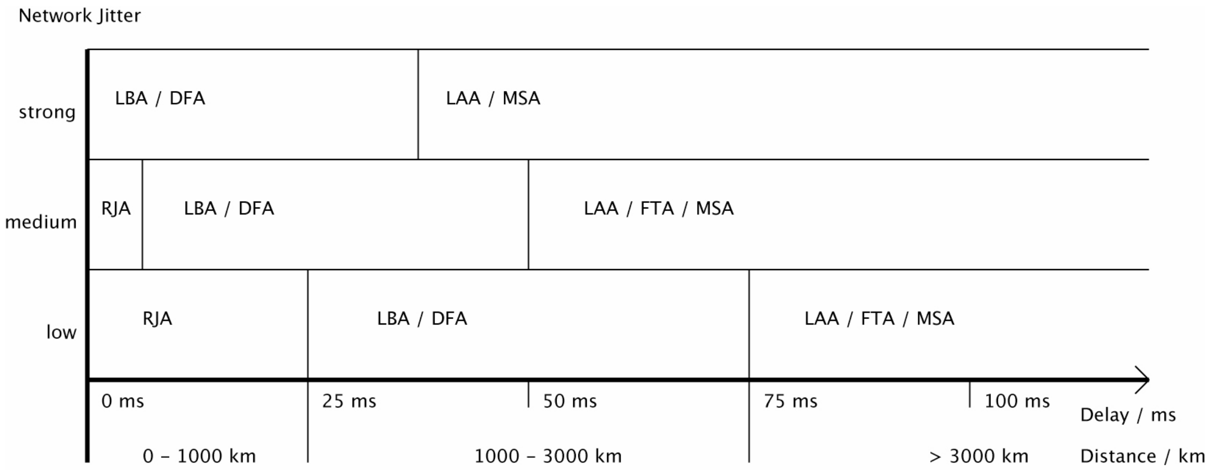


Figure 11: NMP Categories

5 Conclusions and Future Work

NMP has not become a major technology for musical interaction yet. Despite the existence of first commercial products, musicians remain passively in terms of accepting and applying this new approach. Due to the high amount of interdisciplinary knowledge, NMP has so far mainly been used by a small community of experts in IT and music. Apart from audio engineering, network and music skills, the awareness of delay dimensions and their musical consequences is the main basic requirement for a successful network music performance.

In order to give the potential NMP user this ability, our paper describes all categories of delay-influenced musical interaction. Depending on the actual network connection, the user can now consciously apply the suitable category of musical interplay, which allows him to perform under any given network situation. In parallel this gives him awareness of actual possibilities and limitations in his current situation.

In the future we will continue with the evaluation of the mentioned categories with special respect to band sizes of more than two participants. In terms of technical improvements, we will especially investigate in further research for the realistic interaction approach in order to overcome the high challenges of low latency network engineering.

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